A PERFORMANCE ANALYSIS OF THE OPTIMIZED LINK STATE ROUTING PROTOCOL USING VOICE TRAFFIC OVER MOBILE AD HOC NETWORKS

THESIS

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Abstract

Mobile ad hoc networks (MANETs) have grown in popularity over the past decade and are increasingly considered for time-sensitive multimedia applications. The impact of various routing protocols on voice traffic using different IEEE 802.11 extensions has been investigated via analytical models, simulations and experimental test beds. Many studies determined that optimized link state routing (OLSR) is a suitable routing protocol to support voice over internet protocol (VoIP) conversations. This research expands upon this understanding by determining the point at which voice traffic is no longer feasible in an ad hoc environment and determines which audio codec is best suited for MANETS.

The MANET simulation environment is established using OPNET. Varying combinations of workloads are submitted to the MANET to capture voice performance within a stressed environment. Performance metrics are compared against established benchmarks to determine if thresholds for unacceptable voice quality are exceeded.

Performance analysis reveals that VoIP communication using G.711 is not sustainable at walking (1.5 m/s) or jogging (2.5 m/s) speeds when three simultaneous streams are used. Also, G.729a is determined to be the best suited codec for MANETs since it significantly outperforms the other codecs in terms of packet loss and end-to-end delay.

Acknowledgments

I would like to thank the Lord for giving me the opportunity to pursue my passions for His glory. It is only through Him that I have become the person I am today. Without His love, grace and guidance, I would be lost.

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André Wolf

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A PERFORMANCE ANALYSIS OF THE OPTIMIZED LINK STATE ROUTING PROTOCOL USING VOICE TRAFFIC OVER MOBILE AD HOC NETWORKS

I. Introduction

1.1 Background

A mobile ad hoc network (MANET) is a system of wireless mobile nodes that freely and dynamically self-organize into arbitrary and temporary network topologies to allow devices to seamlessly internetwork without any preexisting communication infrastructure [Ily03]. Routing protocols provide a means for wireless nodes to communicate with nodes outside their transmission range by discovering a path between the source and destination. The optimized link state routing (OLSR) protocol is a proactive routing protocol that discovers and maintains routes between source and destination for immediate use while mitigating the effects of overhead control traffic that floods the network. Voice over Internet Protocol (VoIP) is a time-sensitive service that requires consistent performance rates for acceptable voice quality—random packet loss less than 10 percent and end-to-end delay less than 400 msecs. VoIP coder/decoders (codecs), such as G.711, G.726 and G.729a, produce audio payloads of 160, 80 and 20 bytes respectively. VoIP packets include the audio payload created by the codec and 40 bytes of RTP/UDP/IP header information for each packet. The larger audio payloads result in improved voice clarity, but also impact voice performance more significantly if packets are lost or delayed. IEEE 802.11g links between wireless nodes have

communication rates up to 54 Mbps. This research investigates the performance limits of digital voice communications on a MANET in a resource constrained environment.

1.2 Problem Definition and Goal

The growing popularity of MANETs has increased the demand for reliable communication in networks whose topologies change dynamically and where bandwidth is constrained. The use of time-sensitive communications, such as VoIP, has become a prevalent technology found in MANETs. As such, reliable voice communication in MANETs is a valuable capability on the battlefield. The goal of this research is to determine the MANET performance limits at which VoIP becomes unreliable and determine which codec is best suited for MANETs. Specifically, the research stresses the OLSR protocol to determine the effect of using various VoIP codecs in a high mobility and high link utilization environment.

The data collected during simulation is examined to determine the performance impact of VoIP codecs in various MANET environments. The hypothesis of the research is that VoIP codecs with smaller data payloads will outperform VoIP codecs that carry larger data payloads as the MANET becomes stressed with various workloads.

1.3 Approach

Varying combinations of workloads are submitted to the simulated MANET to determine voice performance in a stressed environment. Performance metrics are compared against established benchmarks to determine if thresholds for unacceptable voice quality are exceeded.

1.4 Research Contributions

The impact of various routing protocols on voice traffic using different IEEE 802.11 extensions have been investigated via analytical models, simulations and experimental test beds. Many studies have determined that OLSR is a suitable routing protocol to support VoIP conversations. The goal of this research is to expand upon this understanding by determining the point at which voice traffic is no longer feasible in an ad hoc environment and by determining which audio codec is best suited for an ad hoc network.

1.5 Assumptions / Limitations

The following assumptions and limitations are used in this research. All ad hoc nodes route packets using OLSR, use an IEEE 802.11g wireless interface transmitting at 0.005 W, have infinite battery life, and have infinite IP and WLAN buffers. All traffic is generated from a single node and is sent to other nodes using a random distribution. Each VoIP data stream generates traffic at a rate of 50 packets per second. VoIP packet size of 200, 120 and 60 bytes are used to model G.711, G.726 and G.729a codec respectively. The network area is fixed at 1,000 m by 1,000 m with 30 nodes placed randomly. The random waypoint model is used to simulate travel patterns. The free space propagation model is used to simulate an environment which is outdoors and does not contain obstacles.

1.6 Thesis Organization

This thesis is organized in the following manner. Chapter 2 is a literature review which provides important background information relevant for understanding OLSR,

VoIP and MANETs. Chapter 3 details the methodology used to establish the simulation environment and collect data. Chapter 4 presents the experimental results and analysis of the collected data. Chapter 5 concludes the thesis and summarizes results.

Supplemental information is also provided. Appendix A explains the OPNET simulation setup. Appendix B provides a validation of the simulation model used, and Appendix C lists the raw data collected. Appendix D provides a detailed analysis of the data.

II. Literature Review

2.1 Introduction

This chapter provides background information relevant for voice traffic over a MANET. Section 2.2 gives an overview of MANETs. Routing protocols used in ad hoc networks and examples are discussed in Section 2.3. OLSR is outlined in Section 2.4. Section 2.5 introduces VoIP. Section 2.6 discusses the use of session initiation protocol (SIP) to establish voice conversations, and Section 2.7 discusses the real time protocol (RTP). Section 2.8 discusses voice coding and various codecs. Section 2.9 discusses the IEEE 802.11 standard for wireless local area networks. Section 2.10 describes related work in this research area. Finally, Section 2.11 summarizes the chapter.

2.2 Mobile Ad Hoc Networks

A MANET represents a system of wireless mobile nodes that can freely and dynamically self-organize into arbitrary and temporary network topologies that allow devices to seamlessly internetwork without any preexisting communication infrastructure [Ily03]. MANETs are increasingly considered for complex multimedia applications where quality of service (QoS) attributes must be satisfied as a set of predetermined service requirements [Mis08].

Communication within MANETs differ from that of wired networks in the following aspects [MK05]:

- The wireless communication medium has variable and unpredictable characteristics that cause the signal strength and propagation delay to fluctuate with respect to time and environment.
- The bandwidth availability and battery power are limited in mobile ad hoc networks, thus the algorithms and protocols used need to conserve bandwidth and energy.
- The computing components of wireless devices have low capacity and limited processing power.
- The mobility of nodes creates a continuously changing topology for communications that break routing paths and force new paths to be dynamically formed.
- The wireless medium is a broadcast medium. This results in all nodes hearing packets within the transmission range of a node.

A representative interconnection of nodes in a MANET is shown in Figure 1. Routing paths are established between nodes by using intermediary nodes that are within

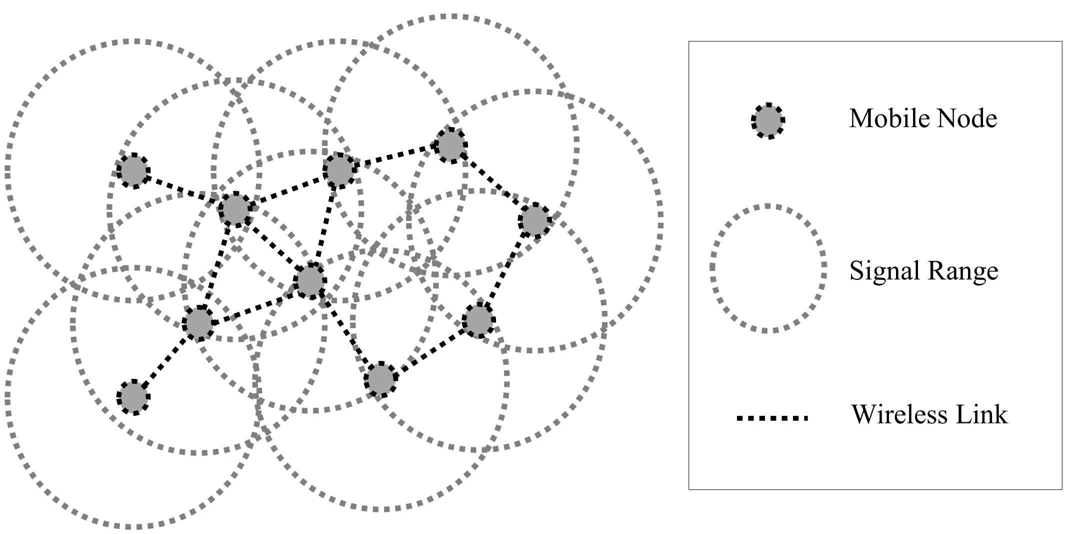


Figure 1: A Mobile Ad Hoc Network [MK05]

overlapping signal ranges of each other. The links between nodes are recalculated by the routing protocol as the topology of the network changes due to mobility.

2.3 Routing Protocols

Routing protocols are generally classified into three categories: proactive, reactive and hybrid. Proactive protocols are table-driven, meaning that they attempt to maintain knowledge of every current route to every other node. Routing information is maintained within each node and routing updates are continuously propagated to other nodes within the MANET. This process results in routes that are instantaneously available [SVV11].

Reactive protocols create routes in an on-demand fashion only when communication is needed. As a result, a route discovery process is performed immediately before a communication link is established between two nodes [SVV11].

Hybrid protocols combine aspects of proactive and reactive protocols where proactive is used by a node to establish routes to its closest neighbors (e.g., within a two-hop radius) and reactive is used by a node if communication is desired with another node that is outside of its closest neighbors radius [Mis08].

The differences between proactive and reactive routing protocols are significant. Since proactive protocols aim to have routes readily available, periodic updates are required to maintain knowledge of the greater network. These periodic updates produce a constant level of overhead even if routes are not being actively used. These periodic updates are considered wasteful if the network does not have a need for such route availability. On the other hand, reactive protocols typically have a smaller amount of overhead since route discovery is only performed when a path between two nodes is

needed. The tradeoff is that packet delay may increase since time is needed to discover routes between the source and destination nodes prior to data transmission [Bou04].

Reactive routing protocols typically function in two phases—path discovery and data forwarding [AY07]. During the path discovery phase, the source node floods the network with a Route Request (RREQ) in order to find a path to the destination. Once the RREQ reaches the destination, a Route Reply (RREP) is unicast back to the initiating source. During this phase, a path between source and destination is established. After path discovery is complete, the data forwarding phase may commence. The intended information is sent to the destination using the path established in the route discovery phase.

There are two ways in which paths can be established and data forwarded in reactive protocols—source routing [JMB01] and node-by-node [PR99].

Source routing, as seen in Figure 2, is the process of establishing a path between source and destination nodes by maintaining the accumulated route between them in the packet header. During the data discovery phase, each node appends its own address to the RREQ and broadcasts the packet to its neighbors until a path to the destination is found. The source, Node A, appends address A to the RREQ and forwards the RREQ to Node B. Node B concatenates address B to the RREQ and forwards the RREQ to Node C. This process continues until the destination, Node E, is reached. Different RREQs sent by the same source node are differentiated using an identification (ID) field. The ID

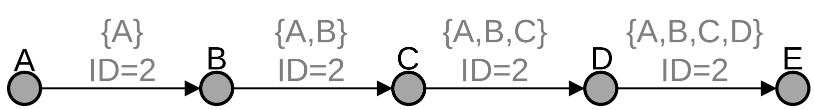
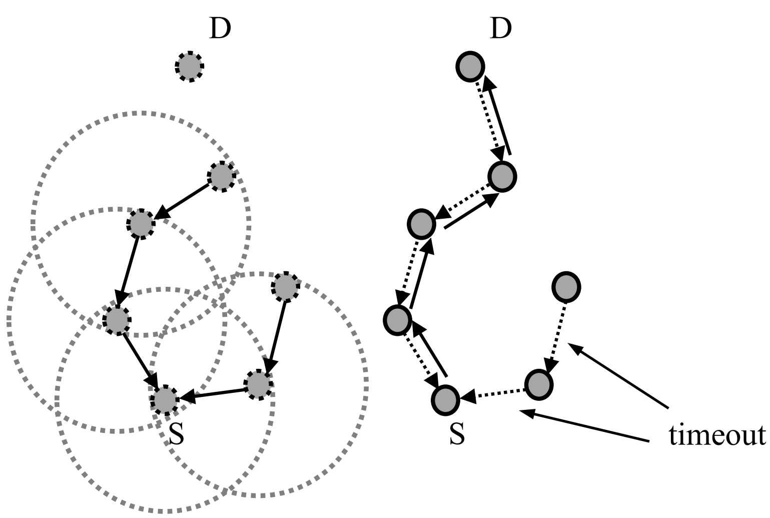


Figure 2: Source Routing [JMB01]

value remains constant for any single RREQ that propagates throughout the network. The path discovered in Figure 2 uses an ID of 2. A RREP is unicast to the source once a path to the destination is discovered. During the data forwarding phase, the source appends this accumulated route (consisting of addresses A, B, C and D) to the header of subsequent packets sent. Intermediate nodes forward packets according to the source route listed in the header of the packet.

Node-by-node routing, as seen in Figure 3, is the process of establishing a path between source and destination, and forwarding packets with route information maintained by intermediate nodes within that path. The route is established by the source, Node S, flooding the network with a RREQ. The source node's neighbors further propagate the RREQ to their neighbors, but maintain a table that points to the one-hop node from which it received the request—the reverse route. Multiple reverse routes that point to Node S are discovered in tandem. When the destination, Node D, receives the RREQ, it responds with a RREP. The RREP typically travels along the path established



(a) Reverse Route (b) Forward Route

Figure 3: Node-by-Node Routing [PR99]

by the reverse route. Concurrently, the intermediate nodes update their routing table so that the path points to the one-hop node in which it received the RREP—the forward route. Reverse routes that do not receive a RREP within a specified amount of time will timeout. In this example, only one reverse route to Node D is discovered.

Proactive protocols typically exchange routing information on a periodic basis to maintain readily available routes throughout the network. The use of RREQ and RREP messages is not needed. Therefore, path discovery is conducted on a constant basis with a neighbor sensing or link freshness mechanism specific to the proactive protocol [PB94].

There are two main routing algorithms used for calculating link costs and propagating routing table updates throughout the network--link-state and distance-vector. These routing algorithms can be used by either the proactive or reactive routing protocols. The link-state routing algorithm assumes that the network topology and all link costs are known globally. An update to the routing table in one node of the network is flooded to all nodes in the network. On the other hand, the distance-vector routing algorithm receives routing updates via broadcasts from only its closest neighbors. Each neighbor that receives such an update recalculates the cost between nodes and sends out its own update. This iterative process continues until all nodes in the network receive the latest information and no other routing updates are broadcast. For both routing algorithms, links with the least cost associated with them are preferred [PB94].

There exist a plethora of ad hoc routing protocols. The following protocols are a representative sample of the variety available.

Ad hoc On-demand Distance-Vector (AODV) [PR99] is a reactive protocol that employs node-by-node routing. AODV requires links to be symmetric. HELLO

messages are used between one-hop neighbors to verify bi-directionality and to detect if links fail. In the event a link fails, an unsolicited RREP is sent to all neighbors to notify them of the detection. The path discovery phase is restarted if a path between nodes is still needed. The HELLO messages have another purpose. HELLO messages also allow nodes to discover new neighbors. Failure to receive a HELLO message indicates that a particular neighbor is inactive.

Dynamic Source Routing (DSR) [JMB01] is another reactive protocol, but employs source routing. DSR is able to operate using symmetrical and asymmetrical links. DSR specifies that each node in the source route is responsible for ensuring that packets travel along the intended link. Otherwise, a route error message is sent to the source node to notify of the broken link. DSR has the ability to update source routing by reusing an active path that is available, as opposed to restarting the path discovery process, as required in AODV.

Destination Sequenced Distance-Vector (DSDV) [PB94] is a proactive routing protocol that utilizes the distance-vector algorithm. Periodic route advertisements are sent to neighbors within transmission range. Destination sequence numbers are used to maintain freshness information of available routes. Nodes update sequence numbers using even numbers. Updates involving odd numbers indicate that the routing update was performed on behalf of the original node. Route updates are sent periodically or when a node is not found, another node updates, or a new node is found.

Zone Routing Protocol (ZRP) [Haa97] incorporates the merits of proactive and reactive routing protocols. The proactive routing zone is comprised of a few mobile

nodes within a defined hop radius (e.g., within a two-hop radius). Reactive routing can be used for nodes outside of the proactive routing zone in an on-demand basis.

The OLSR protocol is used in this study and is discussed in Section 2.4.

2.4 Optimized Link State Routing

OLSR [CJ03] is a proactive table-driven protocol designed specifically for mobile ad hoc networks. OLSR is an optimization of a pure link-state protocol. While the pure link-state protocol declares and floods all neighbor nodes, the OLSR protocol reduces the number of retransmissions that flood a network in two ways. First, the size of the control packets is reduced by declaring only a subset of links with its neighbor nodes. Second, flooding of control traffic is minimized by utilizing only select nodes to defuse messages throughout the network [JMC+01].

OLSR's proactive nature allows routes to be immediately available. For route calculation, each node selects a set of its neighbor nodes which are called multipoint relays (MPR). The MPRs are responsible for forwarding control traffic throughout the network. Overhead information from the flooding of control traffic by using only MPRs significantly reduces the number of retransmissions required to flood a message to all nodes in the network.

Figure 4 illustrates how OLSR floods messages within its two-hop node neighbor set using its one-hop MPR set [Ily03]. The MPRs for Node 1 consist of Nodes 2, 3, 4 and 5. The union of all MPRs consists of the entire set of two-hop nodes. Nodes not selected as a MPR for Node 1 do not forward routing messages to the two-hop nodes.

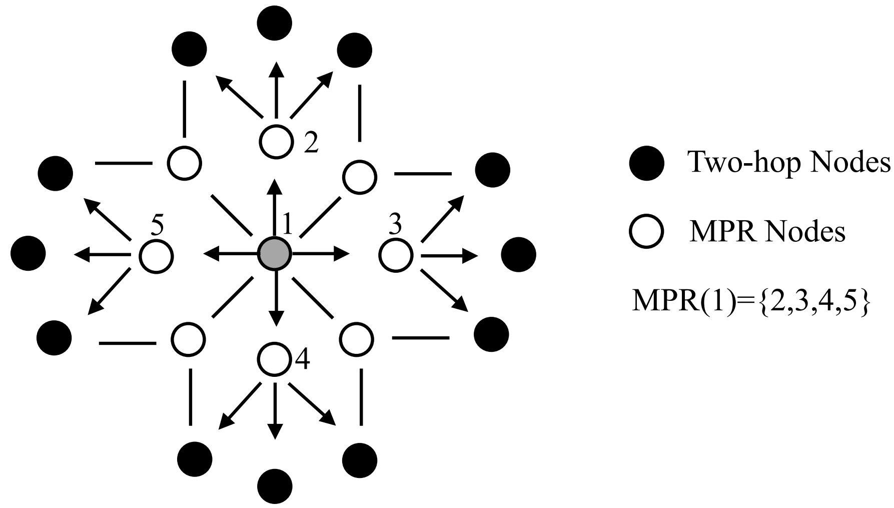


Figure 4: OLSR Flooding Using MPR Nodes [Ily03]

2.4.1 Neighbor Sensing and MPR Selection

The neighbors of Node 1, seen in Figure 4, that are not in the MPR set only read and process the packet, and thus do not retransmit the broadcast message. Each node maintains its own set of MPRs. MPR sets can change over time with the use of HELLO messages [JMC+01].

HELLO messages are transmitted between nodes on a periodic basis and are used to gain information on their neighbors. These messages are only received by one-hop neighbors and are not relayed throughout the network. HELLO messages contain a list of neighbor addresses where there exists a bidirectional link and of which are heard by that particular node. The exchanging of HELLO messages between one-hop nodes allow those nodes to discover nodes that are two-hops away [JMC+01].

Factors such as propagation and mobility may cause some pre-established links to become invalid. Links are only considered valid if they are bidirectional. MPRs for a node are recalculated when a change in the neighborhood is detected when a bidirectional

link fails or when a new neighbor with a bidirectional link is added to the network. Neighbor tables at each node are used to keep record of information about one-hop neighbors, the status of the link with these neighbors, and a list of the two-hop neighbors that one-hop neighbors give them access to. Recalculations of neighbor tables are tracked using sequence numbers that are incremented every time a node selects or updates its MPR set. The use of sequence numbers ensures that nodes maintain the most current routing information. MPR selection at each node is done independently of another. MPR selection is done in a manner where the union of all MPR sets includes the entire two-hop neighbor set. Nodes selected as MPRs maintain a MPR selector set that consists of all nodes that choose it as an MPR [JMC+01].

2.4.2 MPR Information Declaration

Topology Control (TC) messages are used by nodes to broadcast to the network which nodes have selected the sender as a MPR. TC messages are used by each node to build topology tables. Topology tables are used by each node to calculate routing tables. Received TC messages indicate to a node that the destination node can be reached in the final hop through this last-hop node [JMC+01].

The following procedure is executed to record the information in a topology table upon receipt of a TC message [JMC+01]:

- No further processing of this TC message is done and it is silently discarded if there exists some entry in the topology table whose last-hop address corresponds to the originator's address of the TC message and the MPR selector sequence number is greater than the sequence number of the received message.

- The topology entry is removed if there exists some entry in the topology table whose last-hop address corresponds to the originator's address of the TC message and the MPR selector sequence number in that entry is smaller than the sequence number of the received message.
- The holding time of an entry in a topology table is refreshed if there exists some entry in the table whose destination address corresponds to the MPR selector address and the last-hop address of that entry corresponds to the originator address of the TC message. Otherwise, a new topology entry is recorded in the topology table.

2.4.3 Routing Table Calculation

Routing tables allow packets to be routed to their respective destinations and are maintained by each node in the network. A routing table entry consists of a destination address, next-hop address and an estimated distance to destination for each destination in the network for which a route is known [JMC+01].

Routing tables are calculated (or re-calculated) in the following manner [JMC+01]:

1. All entries are removed.
2. New entries are recorded in the table starting with one-hop neighbors as destination nodes. Destination and next-hop address are set to the address of the neighbor and the distance is set to 1.
3. New entries are recorded in the table for destinations that are two hops away. The destination is set to the destination address in the topology table and the

next-hop is set to the next-hop of the route entry whose destination is equal to the desired address; distance is set to 2.

4. The topology table entries which are not used in calculating routes may be removed if there is a need to save memory. Otherwise, these entries may provide multiple routes.

2.5 Voice over Internet Protocol

The evolution from analog to digital communications has made VoIP one of the most ubiquitous Internet applications today. VoIP packetizes phone conversations which allows for transmission using the Internet Protocol (IP) [AKA08]. However, IP is a best-effort service that moves datagrams from sender to receiver as quickly as possible with no guarantee on end-to-end delay [KR10].

VoIP protocols achieve desired functionality by using a signaling protocol (discussed in Section 2.6) for call control, using a media transfer protocol (discussed in Section 2.7) for voice transfer and using voice coding techniques (discussed in Section 2.8) to allow waveforms to be digitized and compressed [AKA08].

2.5.1 VoIP Header

Typical Internet applications use TCP/IP protocols to transfer data across the Internet. However, VoIP packets utilize RTP/UDP/IP, as seen in Figure 5. Audio payloads are time-stamped using RTP which requires a 12 byte header. The resulting segment is then carried by a UDP datagram that requires an 8 byte header. A 20 byte

IPv4 Header (20 bytes)	UDP Header (8 bytes)	RTP Header (12 bytes)	Audio Payload (5-160 bytes, codec-dependent)

Figure 5: VoIP Packet Structure

header is appended to the payload when encapsulated into an IP datagram. The audio payload can vary from 5 bytes to 160 bytes depending on the type of codec used. The result is a VoIP packet that can vary from 45 bytes to 200 bytes per packet where 40 bytes of the packet will always be a header.

2.6 Session Initiation Protocol (SIP)

SIP is a signaling protocol that controls the initiation, modification and termination of interactive multimedia sessions. SIP's peer-to-peer nature allows call routing and session management functions to be distributed across all nodes within a network [DPB+07]. SIP is defined in the Request for Comments (RFC) 2543 and RFC 3261 by the Internet Engineering Task Force (IETF).

A typical SIP call session between two clients is illustrated in Figure 6. A session is initiated by Client A by sending an INVITE request to Client B. An intermediate 100 Trying response is used when the INVITE is sent, but has yet to locate Client B. Client B will be alerted about the request and an interim 180 Ringing response will be sent back to Client A. Client B then sends a 200 OK message upon answering the request. The OK

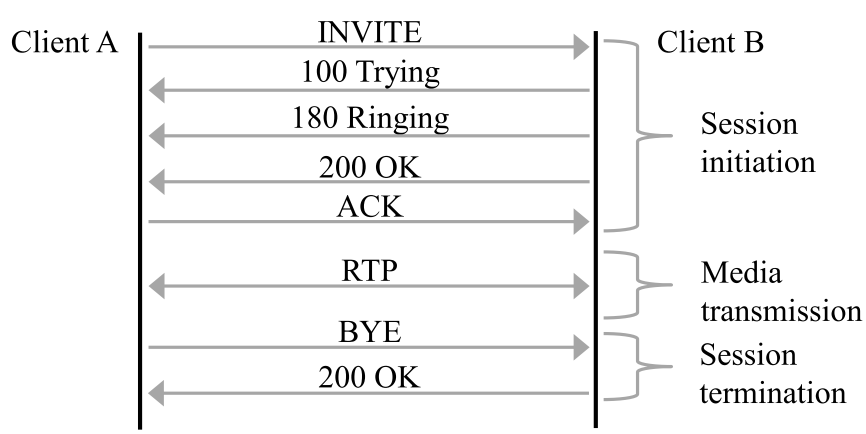


Figure 6: SIP Message Flow [AKA08]

response is acknowledged by Client A by sending an ACK back to Client B. At this point, session initialization is complete [GL05].

The session is established and the two clients begin to exchange data once the INVITE/200/ACK three-way handshake is complete. The specifics of the media transmission are discussed in Section 2.7. Once the conversation is complete, the first client to terminate the conversation sends the BYE message. The other client acknowledges the BYE message by sending back a 200 OK message [GL05].

2.7 Real Time Protocol

RTP is used to deliver voice payloads between applications after the SIP three-way handshake is complete and a session is established between two nodes. RTP is defined in RFC 3550 and is commonly used for transporting video and audio formats. Recall from Figure 5 that when an audio payload is created by an application, it is encapsulated with an RTP packet. The RTP packet is then encapsulated by a UDP header and then the segment is handed to the IP layer for transportation across the network. The IP and UDP headers are used to ensure transport of the packet between the correct source and destination while the RTP header is used by an application to handle the multimedia payload.

2.7.1 RTP Header

The RTP header, illustrated in Figure 7, has four primary fields used by the application: payload type, sequence number, time stamp and synchronization source identifier [KR10].

The payload type field identifies the format of the RTP payload. The payload types available include a wide range of audio and video formats as defined in RFC 3551.

Payload Type (7 bits)	Sequence Number (16 bits)	Time Stamp (32 bits)	Synchronization Source Identifier (32 bits)	Miscellaneous Fields (9 bits)

Figure 7: RTP Header Fields

The payload type used may change during a session [SCF+03]. The sender may want to change the encoding in order to increase the audio quality or to decrease the RTP bit rate [KR10].

The sequence number field is used by the receiving node to detect packet loss and to restore packet sequence. The initial value of the field is random. The sequence number is incremented by one for each RTP data packet that is sent to the destination [SCF+03].

The time stamp field reflects the sampling instant of the first octet in the RTP data packet. The sampling instant must be derived from a clock that increments monotonically and linearly in time. This allows for synchronization between nodes and delay calculations. Time stamps can also be used to reconstruct the timing of a single stream [SCF+03].

The synchronization source (SSRC) identifier is used to identify the source of an RTP stream. This identifier is also chosen randomly. The probability of multiple sources choosing the same SSRC identifier is low since 32-bit identifiers are used. In the event that duplicate SSRC identifier's are detected, the sources involved choose new values for the identifier [SCF+03].

The miscellaneous fields shown in Figure 7 are a collection of five different fields that consist of nine bits. These fields give information on the version of RTP, padding information, header extension, contributing source count and a marker bit [SCF+03].

The RTP Control Protocol (RTCP) is used in conjunction with RTP and is also defined in RFC 3550. The primary function of RTCP is to provide feedback between participants on the quality of the RTP streams. RTCP packets are transmitted by each participant in an RTP session to all other participants in that session on a periodic basis. Information such as sent packet count, packet loss count, and delay are contained in RTCP exchanges between participants [SCF+03].

2.8 Voice Coding

An analog to digital conversion process is required for the digital storage and transmission of any speech or sound signal. The primary factors that influence the quality of digitized speech are sampling rate and quantization level. The sampling rate affects the bandwidth required for transmitting a signal while the quantization level impacts the correctness of the digital representation of the signal. Higher quantization levels also require more bits to be represented per communication symbol. Various codec techniques exist for the digitization of a voice signal [HGP00] [Skl09].

2.8.1 Sampling Theory

As shown in Figure 8, the sampling process can be viewed as multiplying (a) a continuous analog signal by (b) a pulse train of unit amplitude that is spaced by the sampling period. The result is (c) a pulse amplitude modulation (PAM) representation of the original signal [HGP00].

The sampling theorem relies on the fact that the multiplication in the time domain extends to a convolution in the frequency domain. The sampling frequency must be equal to or greater than two times the Shannon-Nyquist criteria. This criterion specifies that the sampling frequency must be at least twice the maximum frequency component of

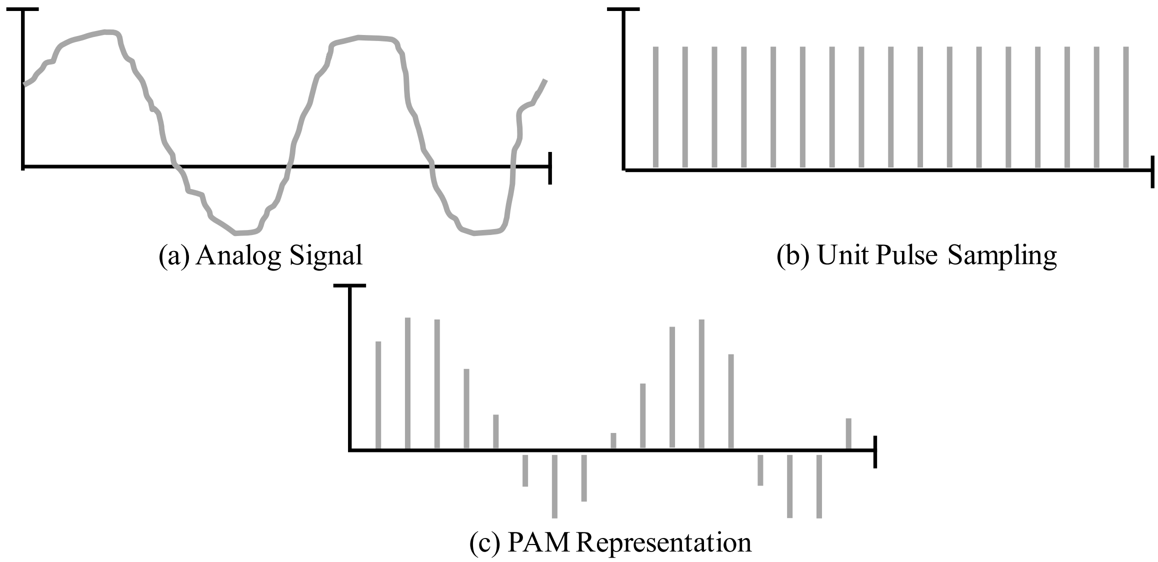


Figure 8: Signal Sampling [HGP00]

the original signal. The value of the sampling frequency determines the transmitted bandwidth which greatly impacts the amount of information transmitted [HGP00].

2.8.2 Quantization

The PAM signal creation process shown in Figure 8 (c) is still analog. Quantization is a process used to convert the amplitude of a PAM signal into rounded values used to create a digital representation of that PAM signal. The noise power increases as the measuring scale gets less precise because a higher level of estimation and rounding is involved. In other words, the audio quality increases as the number of amplitude levels available to estimate and represent the level of a PAM signal increases. There is no chance to improve the quality of an audio signal once this quantization noise is introduced [HGP00].

The quantization process can be linear or non-linear. An illustration of a non-linear quantization process is seen in Figure 9. An analog signal that falls within the

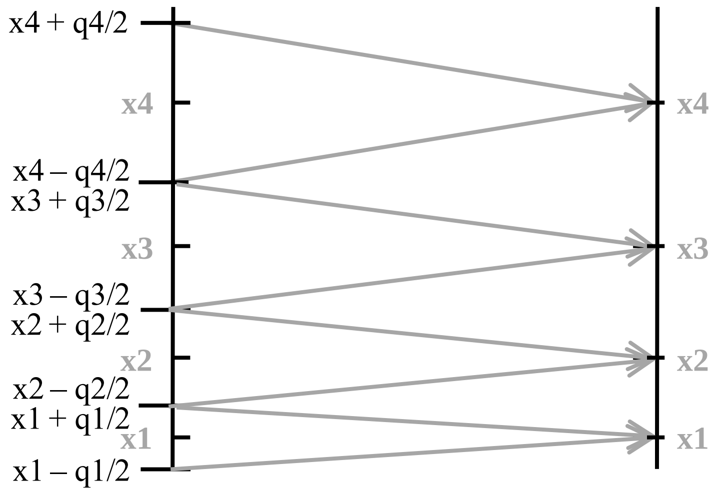


Figure 9: Quantization of Sampled Signal [HGP00]

range of x4 + q4/2 and x4 – q4/2 is assigned the quantized value of x4. In a linear process, the range of the upper and lower ranges for each quintile is the same. Since Figure 9 shows a non-linear process, then the quintile of x3 is defined as any analog value that falls within an even smaller window of x3 + q3/2 and x3 – q3/2. The windows between quintiles get smaller as the quantization value gets lower [HGP00]. This process of non-linear (or non-uniform) quantization is called compression, while at the receiver the reverse of this process is called expansion. Today, most systems use a piecewise linear approximation to the logarithmic compression characteristic [Skl09].

2.8.3 Codec

The human ear can perceive frequencies between 10 and 22000 Hz, although typical phone conversations are limited to frequencies between 200 and 3200 Hz for comprehensible audio communication in the US and Japan [HGP00]. This implies that there is no sense in wasting bits to encode frequencies that are not needed for intelligible phone conversation--the entire range of audio frequencies are not required to be transmitted. High fidelity audio requires higher sampling and bit rates. However,

various compression techniques are available to reduce the bit rates of an audio stream since network throughput may not always allow for the highest fidelity audio to be used. Table 1 shows a sample of the compression techniques available. The bit rate specifies the number of bits per second required to deliver a voice call. The sample size describes the number of bytes captured during the sampling of an analog signal. The packets per second (PPS) represents the packetization rate required to maintain a codec bit rate. The payload size describes how many bytes make up an audio payload of a VoIP packet [San09] [ZBE+07]. Codec's employing a 50 PPS packetization rate generate a new VoIP packet every 0.02 seconds (1 second / 50 PPS) [Cis04].

Table 1: Audio Codecs for Packet Networks [San09] [ZBE+07]

Codec	**Bit Rate** (kbps)	**Sample Size** (bytes)	**PPS**	**Payload Size** (bytes)
G.711	64	80	50	160
G.723.1	6.3	24	33	24
G.723.1	5.3	20	33	20
G.726	32	20	50	80
G.728	16	10	50	60
G.729a	8	10	50	20

2.9 IEEE 802.11

IEEE 802.11 is a standard for wireless local area networks. The purpose of the standard is to foster product compatibility between industry vendors. Vendors design their products to operate within the specifications of the physical and medium access control (MAC) layers [Ily03]. There exist many protocol extensions which are summarized in Table 2.

Table 2: IEEE 802.11 Protocol Extensions [Bro06]

	802.11a	**802.11b**	**802.11g**	**802.11n**
Maximum Data Rate (Mbps)	54	11	54	600
Radio Frequency Band (GHz)	5	2.4	2.4	2.4/5
Number of Spatial Streams	1	1	1	4
Channel Width (MHz)	20	20	20	20/40

2.9.1 MAC Layer

The MAC layer of 802.11 can operate in two basic configurations: infrastructure mode or ad hoc mode. Infrastructure mode allows nodes to communicate between each other by establishing a connection through an access point. In ad hoc mode (or peer-to-peer mode), nodes interact with one another without the support of infrastructure [Ily03].

The MAC layer of 802.11 also specifies the use of the Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA). CSMA/CA helps reduce the number of over-the-air collisions that occur by sensing other carriers before attempting to transmit its own data frame over the wireless medium. If a collision is sensed, then CSMA/CA provides a random back-off period before another transmission is attempted [Ily03]. A sensed collision is typically the result of a hidden node.

Figure 10 shows an example of the hidden node problem that is experienced in wireless networks. Node A and C are outside of each other's transmission range, but

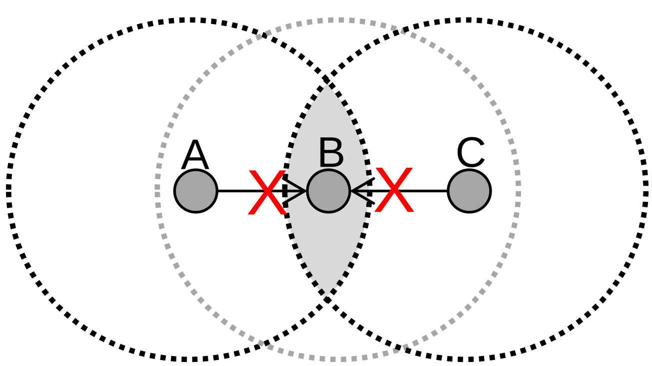


Figure 10: The Hidden Node Problem

both nodes are able to communicate with Node B. In the event that both nodes are trying to transmit a data frame to Node B at the same time, neither is able to successfully communicate with node B because the competing transmissions collide in the overlapping wireless footprint that contains Node B [Gas05].

In the exposed node problem, seen in Figure 11, Node B is trying to communicate with Node A, but is unable to do so because the wireless medium is already occupied by an adjacent node within its wireless footprint—Node A is already communicating with Node 1. Similarly, Node C is communicating with Node 2 which further inhibits Node B's chance to communicate [Gas05].

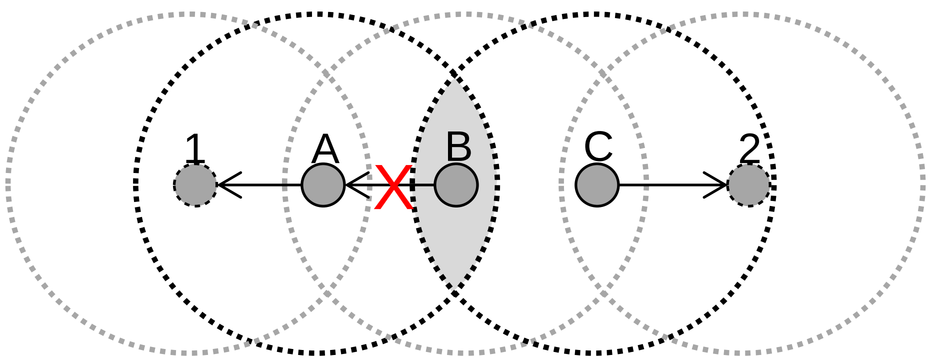


Figure 11: The Exposed Node Problem

2.9.2 Physical Layer

The physical layer of 802.11 specifies the use of various techniques that utilize the same MAC layer, such as Direct Sequence Spread Spectrum (DSSS) and Frequency Hopping Spread Spectrum (FHSS). DSSS utilizes the entire available bandwidth to wirelessly transmit a signal. This transmission technique operates by multiplying a bit stream with higher-frequency signals in order to achieve a spread signal. Spreading is achieved in FHSS by moving the transmitting and receiving mechanisms between 78 narrow channels [Ily03].

2.9.3 802.11 Header

A typical 802.11 header involves 34 bytes of overhead per frame and contains various fields necessary for wireless transmission, as seen in Figure 12. The frame control field is a collection of subfields that include information on protocol version, type, subtype, power management and etc. The duration field specifies the period of time that a channel is reserved for a transmitting node to transmit data and an acknowledgement. The address fields specify MAC addresses involved with the communication between two nodes. The sequence control field is used to distinguish between newly transmitted frames and retransmissions of a previous frame. The payload consists of the information passed between nodes, such as the 5-160 byte VoIP packet depicted in Figure 5. Lastly, a cyclic redundancy check (CRC) field follows the payload and is used to detect bit errors during the wireless transmission [Ava08] [KR10].

Frame Control (2 bytes)	Duration (2 bytes)	Address1 (6 bytes)	Address2 (6 bytes)	Address3 (6 bytes)	Sequence Control (2 bytes)	Address4 (6 bytes)	Payload (0-2312 bytes)	CRC (4 bytes)

Figure 12: An 802.11 Frame

2.10 Related Work

2.10.1 Performance of Various Routing Protocols in MANET Simulation

Sai Anand et al. [SVV11] investigate the performance of VoIP traffic over various routing protocols. Optimized Network Engineering Tools (OPNET) is used to simulate an environment that uses only the G.711 codec for voice coding via IEEE 802.11b and IEEE 802.11g. Each simulation has 20 mobile nodes that are placed randomly with the following attributes: mobility rate of 5 meters per second (mobility model not specified) and buffer size of 256000 bits. Each simulation runs for 1200

seconds. The performance results between the AODV, DSR, OLSR and GRP routing protocols under various VoIP traffic loads are summarized in Table 3.

Table 3: Performance of Protocols [SVV11]

Parameter	Traffic Load	DSR/GRP	AODV	OLSR
Throughput	Low	Poor	Good	Good
	Medium	Poor	Fair	Good
	High	Poor	Poor	Fair
Delay	Low	Poor	Optimum	Optimum
	Medium	Poor	Fair	Optimum
	High	Poor	Poor	Fair

Sai Anand et al. conclude that the OLSR protocol is best suited to handle VoIP traffic in MANETs. OLSR's success is attributed to its proactive nature. The use of MPRs reduce message overhead between nodes. MPR message exchanges allow for route optimization before the need for the exchange of data between nodes.

2.10.2 AODV vs. OLSR Using Various Codecs in Stationary Environment

Armenia et al. [AGL+05] examine the performance of the AODV versus the OLSR protocol using various codecs in OPNET. The authors establish a stationary test bed of four workstations using an IEEE 802.11b network in ad hoc mode. The result of this study is summarized in Table 4, where "O" stands for OLSR is better, "A" stands for AODV is better and "O/A" stands for similar performance between OLSR and AODV. OLSR has better performance in terms of delay and sequence error while AODV

Table 4: Performance of AODV versus OLSR Using Various Codecs [AGL+05]

Codec	Delay	Throughput	Jitter	Seq. Error
G.711-μLaw	O	O/A	A	O
G.711-Alaw	O	O/A	O	O
iLBC-13k3	O	O/A	A	O
iLBC-15k2	O	O/A	A	A
GSM-06.10	O	O/A	A	O
MS-GSM	O	O/A	A	A

outperforms in the jitter category. Similar results for throughput are produced for both routing protocols.

2.10.3 Performance of Various Codecs in MANET Simulation

Islam et al. [IIA+10] measure the impact that the number of nodes in a MANET have on the performance of voice codecs in a simulation environment. Seven different voice codecs are compared as the number of mobile nodes increases from a medium density of 25 mobile nodes to a heavy density of 50 mobile nodes. The authors conclude that G.711 is best suited for medium node density with an average delay of 0.097 seconds while the Global System for Mobile Communications Enhanced Full Rate (GSM-EFR) codec outperforms the other codecs in heavy node density situations with an average delay of 0.14 seconds.

2.10.4 MANET Performance in a VoIP Context

Thibodeau et al. [TYH06] simulate the effects of node density, number of data streams and route length on the performance of VoIP traffic using only the AODV routing protocol. Network Simulator 2 (ns-2) is used as the simulation tool to evaluate delay, jitter, and the duration and quantity of interruptions. The research shows that varying node density from 30 to 90 has no significant impact on performance of VoIP traffic. However, an increase in route length and number of streams degrades the performance of voice traffic in terms of delay and jitter.

This study reports an issue in the MAC layer of the 802.11 standard while using AODV. An error is returned to the routing protocol when a link is falsely reported as unavailable. The false reporting causes the AODV routing protocol to repair the route or search for a new route. Chaudet et al. [CDL05] report that the false detection of lost links

is caused by the well known issues of the hidden/exposed node problem and 802.11 MAC layer unfairness.

2.10.5 VoIP Performance Analysis of OLSR MANET

Santos [San09] analyzes the impact of node density, mobility and number of data streams on the performance of voice traffic. The simulation environment is implemented using OPNET with IEEE 802.11g links between nodes. One node is used to stream G.711 VoIP traffic to random destinations within the MANET. Mobile nodes employ the random waypoint model when in motion. The performance metrics used to measure impact on voice traffic include delay and packet loss.

Santos concludes that OLSR is a suitable routing protocol to support VoIP traffic. The impact of node density, mobility and number of data streams in this investigation shows that delay and packet loss are within acceptable levels to sustain VoIP conversations. The research does not include the impact of increasing the number of streams to the point of IEEE 802.11g link saturation or on the impact of increasing the mobility beyond walking speeds.

2.10.6 Mobility Models for Ad Hoc Network Research

Camp et al. [CBD02] compare the use of seven different mobility models in the performance evaluation of a protocol for MANETs. For brevity, only the Random Waypoint Model (RWM) is discussed here. The RWM is widely used in many prominent simulations of MANETs. The authors observed that the RWM appears to create realistic mobility patterns for the way people move in an enclosed environment. This model also appears to be stable at high speeds where pause times between movements are over 20 seconds.

The RWM consists of nodes that change speed and direction between pause times. This model operates by the node selecting a destination with a given environment and then moving toward that destination in a straight line with a speed that is selected uniformly between a minimum and maximum value. The mobile node pauses for a specified amount of time once it reaches the destination. After the pause period, a new destination and speed is selected [CBD02]. Figure 13 shows the movement pattern of a mobile node using the RWM.

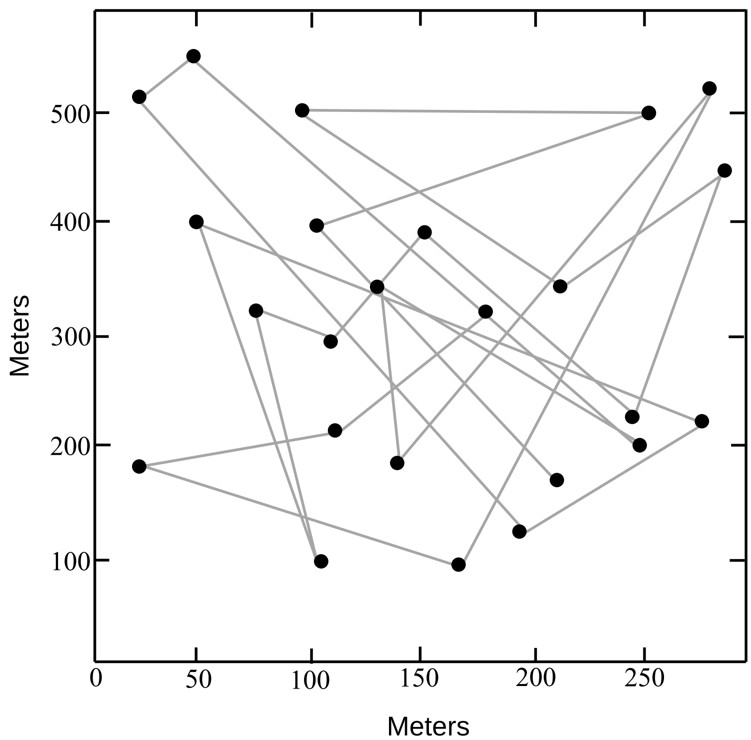


Figure 13: Travel Pattern Using Random Waypoint Model [CBD02]

Camp et al. suggest that there is an initialization issue with the RWM. Discarding the initial 1000 seconds of simulation for each trial ensures that each trial has a random initial configuration. Additionally, the use of faster speeds for mobile nodes requires a smaller amount of simulation time to be discarded because a random initialized configuration is reached faster [CBD02].

2.10.7 Throughput Performance of Saturated 802.11g Networks

Dao et al. [DM07] discuss the analytical and simulated performance of MANETs using saturated IEEE 802.11g links between nodes. Saturated throughput between nodes occurs when every station has packets ready to transmit whenever the channel is available. OPNET simulations and analytical results show that throughput decreases between nodes as the number of nodes in the MANET increases when all mobile nodes transmit at the maximum IEEE 802.11g physical data rate of 54 Mbps.

The impact of varying the contention window on the throughput of a saturated 802.11g network is also studied by Dao et al. The authors conclude that large contention windows are best suited to maximize voice capacity [DM07]. No mention is made of the impact on the performance of voice traffic.

2.10.8 Performance Evaluation of Ad Hoc Routing Protocols

Boukerche [Bou04] investigates the performance of four routing protocols—three reactive and one proactive. This study analyzes the impact of varying node density, traffic load and mobility on throughput, end-to-end delay and routing overhead. Simulations are conducted in ns-2. Node density is varied from 50 to 100 nodes, while traffic loads vary from 10 to 30 connections. Mobility varies by incrementally increasing the pause time of the random waypoint model from 0 secs to the max time of simulation. Therefore, lower mobility is experienced for higher pause times. Scenarios involving node densities of 50 have a simulation time of 900 sec while scenarios involving node densities of 100 have simulation times of 500 secs.

Bourkerche’s research concludes that DSR has the highest throughput due to the efficiencies of source routing. AODV has the shortest end-to-end delay due to the ability

of hop-by-hop routing to adapt to changing network topologies. Lastly, DSR experiences the least amount of network overhead since there is no use of HELLO messages. DSDV is the only proactive protocol in this study and performs poorly in most situations.

2.10.9 Performance of Ad Hoc Routing Protocols in IEEE 802.11

Performance comparisons between proactive and reactive protocols are investigated by simulating constant-bit-rate (CBR) data traffic in an ad hoc network using ns-2. Putta et al. [PPR+10] state that the CBR source is modeled as generating packets at with a fixed packet size of 512 bytes (codec not specified). The bit rate is not varied. Data traffic that is not time-sensitive is modeled as file transfer protocol (FTP) traffic. OLSR is used to represent the proactive protocol while AODV and DSR are used to represent reactive protocols. The simulation varies traffic loads and mobility speeds.

Putta et al. conclude that proactive protocols are better suited for time-sensitive traffic while reactive protocols are better adapted for data services without strict timing requirements. OLSR performs better in terms of delay than reactive protocols when the network experiences heavier traffic loads. OLSR was able to preserve lower end-to-end delays by having routes readily available, but the resulting routing traffic overhead was higher in order to maintain routes within the network. AODV and DSR are able to provide 80% packet delivery for a heavy FTP traffic load even as the mean end-to-end delay increases.

2.10.10 Reactive versus Proactive Routing Protocols

Mbarushimana et al. [MS07] also compare proactive and reactive routing protocols in MANETs. AODV, DSR and OLSR are simulated in OPNET, while varying the number of data streams, stream intensities, node density and mobility.

Ten CBR streams with varying intensities from 12.5 to 150 Kbps are used to represent time-sensitive media traffic. The simulation results demonstrate that OLSR is the superior protocol when traffic intensity increases–maintaining 1.3 Mbps throughput while at high network intensity. The effect of the number of data streams is evaluated by varying the number of traffic sources from 5 to 30. OLSR results in the least amount of delay when compared to other protocols. The network size is varied by increasing node density from 25 to 100 nodes within the simulation area. The end-to-end delay increase experienced by all routing protocols is relatively low. However, the routing load increases in a linear fashion for reactive protocols as the network size grows; whereas, proactive routing protocols experience an exponential growth in routing overhead as the network size increased. Lastly, mobility varied from zero to 20 m/s. AODV outperforms DSR as mobility increases, but OLSR outperforms both. OLSR's superiority is attributed to the fact that it is able to detect failed links sooner, thus resulting in fewer dropped packets.

Mbarushimana et al. conclude that proactive routing protocols are best suited to route time-sensitive traffic since OLSR experiences lower end-to-end delay and higher delivery ratios. Reactive routing protocol's negative performance is attributed to the amount of time it takes to establish a route—buffers are prone to overflow while the protocol is still calculating a route to the destination.

2.11 Conclusion

The impact of multiple routing protocols on voice traffic using various IEEE 802.11 extensions have been repeatedly investigated via analytical models, simulations and experimental test beds. Many of these studies indicate that OLSR is a suitable

routing protocol to support VoIP conversations. The goal of this research is to expand upon this understanding by determining the point at which voice traffic is no longer feasible in an ad hoc environment and by determining which audio codec is best suited for an ad hoc network.

III. Methodology

3.1 Introduction

This chapter details the methodology used to establish the simulation environment and collect data. Section 3.2 provides the problem definition. Section 3.3 defines the system boundaries, and Section 3.4 details the system services. Section 3.5 describes the workload submitted to the system. Section 3.6 explains the performance metrics used in this study. Section 3.7 describes system parameters; Section 3.8 lists the factors used in this study. Section 3.9 describes the evaluation technique used. Section 3.10 explains the experimental design, and Section 3.11 summarizes this chapter.

3.2 Problem Definition

This section describes the specific goals of the research along with a hypothesis of the expected results. The approach describes how the hypothesis is tested in support of the research goals.

3.2.1 Goals and Hypothesis

The goal of this research is to determine the point at which voice traffic is no longer feasible in an ad hoc environment and determine which audio codec is best suited for an ad hoc network.

This research stresses the OLSR protocol to determine the effect of using various VoIP codecs in a high mobility and high link utilization environment. The data collected during simulation is examined to determine the performance impact of VoIP codecs in various MANET environments. The hypothesis of this research is that VoIP codecs with

smaller data payloads will outperform VoIP codecs that produce larger data payloads as the MANET becomes stressed with various workloads.

3.2.2 Approach

Varying combinations of workloads are submitted to the simulated MANET to determine voice performance within a stressed environment. Performance metrics are compared against established benchmarks to determine if thresholds for unacceptable voice quality are exceeded.

3.3 System Boundaries

The System Under Test (SUT) for this research is the Voice Over MANET System (VOMS). Figure 14 shows that the VOMS consists of four major components—the wireless protocol, routing protocol, ad hoc nodes and ad hoc network.

3.3.1 Wireless Protocol

The physical and MAC layers of a wireless protocol provide a means for two or more nodes to communicate with each other if a direct link exists. The IEEE 802.11g

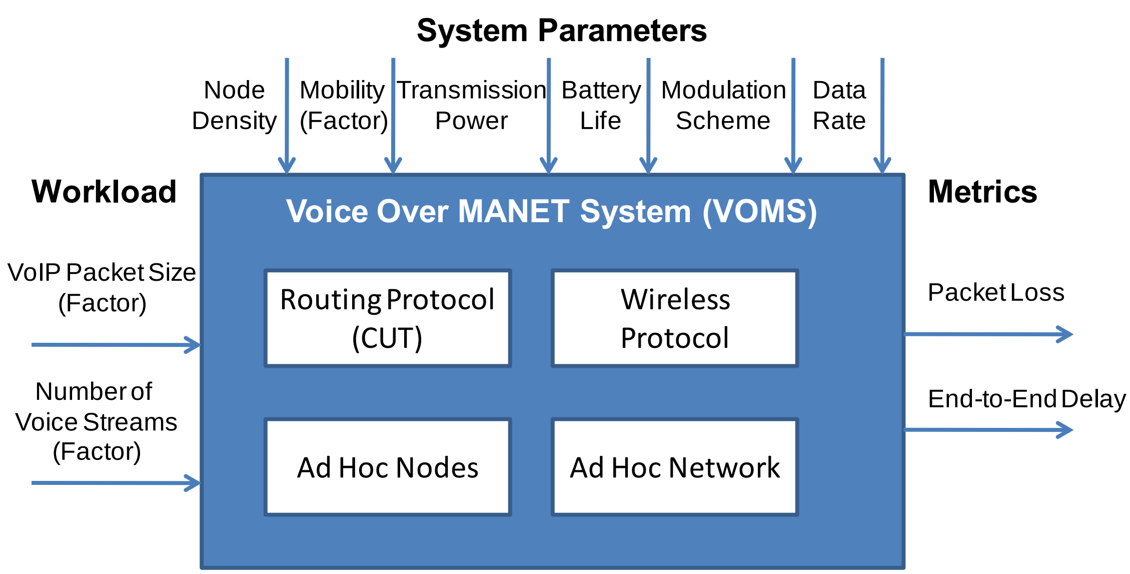


Figure 14: The System Under Test (SUT)

wireless protocol is used for this study. Section 2.9 describes the specifications of IEEE 802.11g.

3.3.2 Routing Protocol (CUT)

Routing protocols provide multi-hop communication between source and destination nodes when there is no direct links between these nodes. OLSR is a proactive routing protocol that establishes and maintains paths between all nodes within VOMS. OLSR is the component under test (CUT).

3.3.3 Ad Hoc Nodes

Ad hoc nodes are the component of VOMS that initiate, receive and maintain voice communications. Thus, ad hoc nodes function as a source, destination or router. As a source, the node generates VoIP packets and wirelessly transmits the packets within the VOMS as a means to reach its intended target. As a destination, the node receives VoIP packets and decodes them for users. As a router, the node maintains path information to all other nodes in the VOMS. These nodes serve as intermediaries that ensure data delivery between source and destination.

3.3.4 Ad Hoc Networks

The ad hoc network operates without the support of a fixed infrastructure. It consists of all ad hoc nodes in the network. Ad hoc nodes maintain communication between each other through the use of wireless links and routing protocols. The interaction between ad hoc nodes in the network has a significant impact on the performance of voice traffic between source and destination nodes.

3.4 System Services

The service VOMS provides is VoIP over a MANET. VOMS accepts VoIP data streams from a source and transmits the stream to a destination. The routing protocol (CUT) is responsible for proactively discovering and maintaining routes between nodes, while the wireless protocol actually transports data between nodes. The possible outcomes of the VOMS, as seen in Figure 15, are:

- The packet is received with no errors and no re-routing required
- The packet is received, but a new route is required and is discovered
- The packet is received, but a new route is required and is not discovered
- The packet is not received due to invalid route
- The packet is not received due to network error

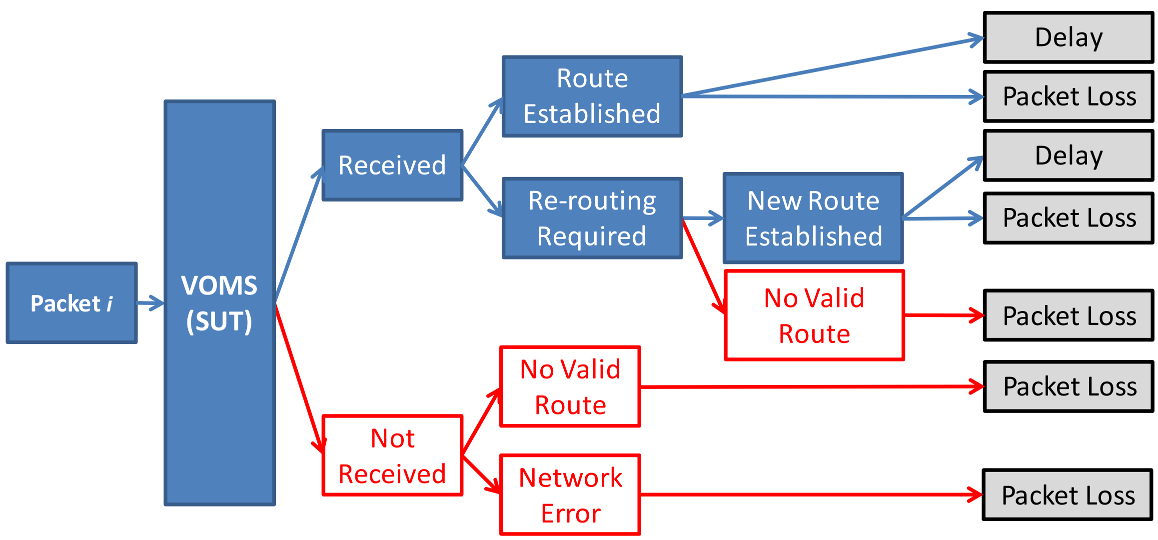


Figure 15: Outcome of Service Request

3.5 Workload

The workload of the VOMS consists of the voice payload size and number of voice streams submitted to the SUT. Voice streams are generated from a single source and transmitted to random nodes throughout the network.

3.5.1 VoIP Packet Size (Factor)

Voice streams consist of a series of consecutively generated VoIP packets. As discussed in Section 2.5.1, VoIP packets consist of 40 bytes of header and a payload size that is dependent on the codec used. The payload size is varied by using the G.711, G.726 and G.729a codecs. When varied, the size of the VoIP packet affects the performance of the system [AGL+05] [SVV11]. Section 3.8 discusses codec as a factor in this study.

3.5.2 Number of Voice Streams (Factor)

The number of voice streams generated by the source affects system performance [San09] [TYH06]. Increasing the number of voice streams increases the utilization of the wireless links. Wireless link characteristics of the system are described in more detail in Section 2.9. Section 3.8 discusses the number of voice streams as a factor in this study.

3.6 Performance Metrics

VoIP is a time-sensitive service. However, since the IP protocol is a best-effort service, VoIP packets are susceptible to packet loss and end-to-end delay.

3.6.1 Packet Loss

Packet loss occurs when VoIP packets fail to arrive at the receiving application and are discarded. This can be due to node buffers becoming full such that they cannot accept any additional IP datagrams. Packet loss can be reduced by using TCP; however, any retransmission of packets increases end-to-end delay to an unacceptable level for voice quality. Random packet loss is acceptable, but rates in excess of 10 percent are not tolerable [ITU03] [KR10]. Packet loss can be estimated by a packet delivery ratio. This is

the ratio of the total number of data packets successfully delivered to the destination to the total number of data packets generated by the source [ZBE+07].

3.6.2 End-to-End Delay

End-to-end delay is the accumulation of transmission, processing and queuing delays at the nodes; propagation delays in the wireless links; and receiving node processing delays. The human ear usually does not perceive end-to-end delays smaller than 150 msecs. While end-to-end delay between 150 and 400 msecs is considered acceptable, any delay in excess of 400 msecs severely impacts voice quality because these packets are considered lost [ITU03] [KR10]. Network delay can be calculated by averaging the end-to-end delay of packets successfully delivered from the source to the application at the destination node [ZBE+07].

3.7 System Parameters

The parameters described below affect performance of the VOMS.

3.7.1 Node Density

Node density is the number of ad hoc nodes in the MANET for a fixed area. Varying levels of node density has an impact on the performance of the system [IIA+10] [San09] [TYH06]. The node density is fixed at 30 nodes within a 1,000 m by 1,000 m area.

3.7.2 Mobility (Factor)

Mobility describes the movement of ad hoc nodes in VOMS. There are many mobility models, as MANET performance is very sensitive to mobility models [CBD02]. The RWM is used to model each node. Two levels of mobility are considered using the

RWM: walking and jogging speed of nodes. Section 3.8 discusses mobility as a factor for this study.

3.7.3 Transmission Power

The maximum allowed output power of intentional radiators operating within the 2400-2483.5 MHz frequency band used by IEEE 802.11g is limited to 4 Watt by the Federal Communications Commission [FCC11]. However, most mobile devices use significantly less power to preserve battery life. Transmission power of a mobile node can significantly affect broadcast range. Transmission power for this study is fixed at 0.005 W.

3.7.4 Battery Life

Battery life in an ad hoc node impacts the amount of time that a node can actively participate in a MANET. Ad hoc nodes are expected to have sufficiently long battery life to sustain communication. Battery life is not a factor in this study; thus, it is assumed that ad hoc nodes will not deplete their batteries.

3.7.5 Modulation Scheme

There are several modulation schemes for IEEE 802.11g. All ad hoc nodes have the same modulation scheme to maintain interoperability between wireless links. Failure to have the same modulation scheme will result in failure to communicate. All nodes will communicate using the Extended Rate PHY (802.11g) model.

3.7.6 Data Rate

IEEE 802.11g supports several data rates. Data rates are the speed in which information is transmitted. All ad hoc nodes operate at a data rate of 54 Mbps. The free

space propagation model is used to simulate an environment that is outdoors and does not contain obstacles.

3.8 Factors

Factors are parameters that are varied between levels and submitted to the SUT. The results of varying these factors are measured by the performance metrics. A summary of factors and levels used in this study is summarized in Table 5.

Table 5: VOMS Factors and Levels

Factors	Units	Levels
VoIP Packet Size	Bytes	200, 120, 60
Number of Voice Streams	Streams	1, 2, 3
Mobility	Meters Per Second	1.5, 2.5

3.8.1 VoIP Packet Size

The G.711, G.726 and G.729a codecs produce an audio payload of 160, 80 and 20 bytes respectively. All audio payloads are encapsulated by a 40 byte RTP/UDP/IP header prior to being transmitted. Therefore, the factor levels are 200, 120 and 60 bytes. Codecs resulting in smaller audio payloads should outperform other codecs in terms of packet loss and delay as the VOMS becomes more stressed since each packet carries less data.

3.8.2 Number of Voice Streams

Voice streams originate from the source node and consist of VoIP packets. The source node randomly determines the destination of each voice stream at the beginning of the simulation and the destination remains the same for the duration of the simulation. The factor levels are 1, 2 and 3 streams. Increasing the number of voice streams will increase the utilization of wireless links between ad hoc nodes. The increased utilization

of wireless links should increase packet loss and delay of VOMS since each link between nodes is limited by the data rate of the wireless link.

3.8.3 Mobility

The factor levels are 1.5 and 2.5 meters per second (m/s). Each ad hoc node is modeled in OPNET using the RWM. Walking speed is set to 1.5 m/s while 2.5 m/s emulates jogging speed. As discussed in Section 2.10.6, the RWM is stable for high speeds when the pause time is over 20 seconds [CBD02]; however, the pause time for this study is set to a constant time of 100 seconds for added stability and to emulate a realistic resting period between distances traveled. Increasing the speed should cause packet loss and delay to increase since paths between source and destination are more likely to be broken. Broken routes force the recalculations of new routes by the routing protocol.

3.9 Evaluation Technique

3.9.1 Simulation

It is expensive and impractical to build an actual MANET for obtaining performance data in this study. Therefore, the evaluation technique for this study is simulation.

Simulations are run using OPNET Modeler Educational Version 15.0 A PL1 (Build 8168 32-bit) on a Dell Precision T7500 computer running Windows 7 Enterprise version 6.1.7601 (Build 7601 64-bit). Ad hoc nodes are placed randomly within a 1,000 m by 1,000 m area using the manet_station node model. Table 6 lists the configuration of the ad hoc nodes using OPNET's wireless suite.

VoIP traffic is introduced into VOMS with a constant packet size that is dependent on the codec used and a constant inter-arrival time of 0.02 sec starting at 0.0 sec with a stop time being the end of simulation. Section 2.8.3 explains codec requirements.

Each node is configured to implement the OLSR protocol. Table 7 shows the OLSR configuration used for all simulations.

Table 6: OPNET Ad Hoc Node Wireless Suite Configuration

Parameter	Setting
Routing Protocol	OLSR
Area	1,000 m by 1,000 m
Node Placement	Random
Power	0.005 W
MAC Layer	802.11g
Data Rate	54 Mbps
PCF/HCF	Disabled
IP / WLAN Buffers	Infinite
Propagation Model	Free Space Model
Mobility	RWM @ 1.5 m/s, RWM @ 2.5 m/s

Table 7: OPNET OLSR Protocol Configuration

Parameter	Setting
Willingness	Default
HELLO interval	2 sec
TC interval	5 sec
Neighbor hold time	6 sec
Topology hold time	15 sec
Duplicate message hold time	30 sec
Internet Protocol	IPv4

3.9.2 Validation

The wireless OPNET model is validated in four ways. This ensures that the routing protocol, wireless protocol and mobility model behave as expected and faithfully

model the real world. Appendix B explains the validation setup and process in more detail.

The IEEE 802.11g wireless protocol is validated by observing the amount of packet loss experienced by varying the distances between sender and receiver nodes. The same number of packets is sent for each distance. Table 8 shows that as the two stationary ad hoc nodes are separated in distance, the amount of packet loss experienced increases. A real world transmission range of 91 m is expected [Bro03]; however, an effective transmission range of 371 m is available in OPNET when using the free space propagation model [GMC08]. Table 8 also shows that nodes communicating at distances in excess of 373 m experience severe packet loss. Recall that VoIP communication is not sustainable with random packet loss in excess of 10%.

Table 8: Wireless Link Validation Results

Distance (meters)	**Distance** (feet)	**Packets Sent** (packets)	**Packets Received** (packets)	**Observed Packet Loss** (%)
300.00	984.25	86300	86300	0.00%
350.00	1148.29	86300	86260	0.05%
360.00	1181.10	86300	85827	0.55%
370.00	1213.91	86300	82723	4.14%
371.00	1217.19	86300	82723	4.14%
372.00	1220.47	86300	82723	4.14%
373.00	1223.75	86300	82723	4.14%
374.00	1227.03	86300	70716	18.06%
375.00	1230.31	86300	70716	18.06%
380.00	1246.72	86300	70716	18.06%
390.00	1279.53	86300	45920	46.79%
400.00	1312.34	86300	19674	77.20%
410.00	1345.14	86300	5142	94.04%
420.00	1377.95	86300	796	99.08%
430.00	1410.76	86300	796	99.08%
440.00	1443.57	86300	62	99.93%
450.00	1476.38	86300	4	100.00%

A node's routing protocol should route packets to a destination outside of its broadcast radius. This is accomplished with five nodes: a source node, three intermediate nodes and a destination node. All ad hoc nodes are stationary. The source and destination node are outside of each other's broadcast range, but both are able to communicate with the intermediate nodes between them. All packets sent by the source node are received by the intermediate nodes and forwarded to the destination node. Each packet sent travels a total of four hops to the destination.

The OLSR routing protocol is further validated with the use of three nodes. The source and destination nodes are outside of each other's broadcast range and are stationary. The intermediate node is initially outside of the source and destination nodes broadcast range, but then moves within range of both nodes. The intermediate node exchanges routing information with the source and destination nodes when within range. The sender node then selects the intermediate node as a MPR and forwards packets to the destination.

The mobility model is validated by observation. Ad hoc nodes exhibit movement pattern consistent with the RWM. Each node randomly selects a destination and travels to the destination at a fix speed. Once the destination is reached, the node pauses for 100 seconds, selects a new destination and then travels to the new destination at the same fixed speed. All nodes stay within the bounds of the 1,000 m by 1,000 m simulation area.

3.10 Experimental Design

A full factorial design is used to evaluate the interaction between factors. The factors include VoIP packet size, number of voice streams and mobility with 3, 3 and 2 levels respectively. This results in 3 x 3 x 2 = 18 scenarios. Since variability is expected

to be high, each scenario will be repeated 30 times with different seed values, and a 90% confidence level is used. This results in 18 x 30 = 540 total experiments. The VOMS for each experiment reaches steady state at different times. Each experiment runs for 2 hours of simulation time in order to ensure a steady state is reached for all experiments. Table 9 lists the seed values used for each repetition in OPNET.

Table 9: OPNET Seed Values

149	953	811	1019	37	17	397	683	277	977
2833	929	107	421	463	1511	809	449	1109	601
504	311	1604	542	78	1819	357	737	698	892

3.11 Methodology Summary

This study investigates the point at which voice traffic is no longer feasible in an ad hoc environment and determines which audio codec is best suited for a MANET. The system under test is the Voice Over MANET System, while the OLSR routing protocol is the component under test. Packet loss and end-to-end delay metrics determine the routing protocols ability to transport time-sensitive VoIP packets in a stressed environment. OPNET simulation is the evaluation technique used in this study. There are a total of 18 scenarios that are repeated 30 times and metrics are reported with a 90% confidence level.

IV. Results and Analysis

4.1 Introduction

This chapter presents analysis of the simulation results collected from OPNET. Analysis is conducted using the R statistical package version 2.13.1 [ISM11]. Section 4.2 discusses exploratory data analysis. Section 4.3 examines the impact of VoIP packet size on end-to-end delay and packet loss. Section 4.4 details the impact of the number of voice streams on end-to-end delay and packet loss. Section 4.5 investigates the impact of mobility on end-to-end delay and packet loss. Section 4.6 utilizes the analysis of variance to show which factor has the most significant impact. Section 4.7 gives an interpretive analysis of the performance metrics observed. The chapter is summarized in Section 4.8. Raw data and analysis can be found in Appendix C and Appendix D respectively.

The goal of this research is to determine the point at which voice traffic is no longer feasible in a stressed ad hoc environment and determine which audio codec is best suited for a MANET.

4.2 Exploratory Data Analysis

This section presents the data collected from simulation to determine its underlying structure. This allows for maximum insight into the data prior to formal analysis.

4.2.1 Data Organization

The data collected in this study consists of 18 scenarios. Each scenario is given a scenario number and scenario name which is used for labeling end-to-end delay and

packet loss data collected, as seen in Table 10. The naming convention used for the scenario name follows: “Mobility”_“# of Streams”_“VoIP Packet Size”.

Table 10: Data Organization

Scenario #	Scenario Name	Mobility (m/s)	# of Streams	VoIP Packet Size (bytes)
1	2.5_3_200	2.5	3	200
2	2.5_3_120	2.5	3	120
3	2.5_3_60	2.5	3	60
4	2.5_2_200	2.5	2	200
5	2.5_2_120	2.5	2	120
6	2.5_2_60	2.5	2	60
7	2.5_1_200	2.5	1	200
8	2.5_1_120	2.5	1	120
9	2.5_1_60	2.5	1	60
10	1.5_3_200	1.5	3	200
11	1.5_3_120	1.5	3	120
12	1.5_3_60	1.5	3	60
13	1.5_2_200	1.5	2	200
14	1.5_2_120	1.5	2	120
15	1.5_2_60	1.5	2	60
16	1.5_1_200	1.5	1	200
17	1.5_1_120	1.5	1	120
18	1.5_1_60	1.5	1	60

4.2.2 Summary Statistics

Summary statistics for all 18 scenarios are presented in Table 11. The raw data used in this study is found in Appendix C for end-to-end delay and packet loss. Recall that each scenario is repeated thirty times with different seeds for each repetition; however, only the repetitions which reach a steady state for end-to-end delay and packet loss are retained. Repetitions that do not reach steady state are considered unstable systems and are deemed invalid. The number of retained repetitions for each scenario is recorded in Table 11 as *n*. Scenario 1 using seed values of 277 and 504, and Scenario 10

using the seed value of 277 are the only three repetitions that do not reach steady state for the metrics of interest and thus are not retained.

The mean, median and standard deviation for end-to-end delay and packet loss are also shown in Table 11. The end-to-end delay summary statistics suggests that delay is not only greater for the most stressed scenarios for jogging and walking, Scenario 1 and Scenario 10 respectively, but variability in delay is also significantly greater. The packet loss summary statistics suggest that scenarios involving 200 byte VoIP packets (G.711), Scenarios 1, 4, 7, 10, 13 and 16, experience greater packet loss and variability.

Failures are quantified as repetitions that experience an average end-to-end delay greater than 400 ms or packet loss greater than 10%. Table 11 shows five scenarios that experience repetitions (of those retained) that meet or exceed the failure threshold defined for either end-to-end delay or packet loss—Scenarios 1, 2, 3, 10 and 11.

Table 11: Summary Statistics

Scenario #	Scenario Name	n	End-to-End Delay (sec)			Packet Loss (%)			Failures
			Mean	Median	St Dev	Mean	Median	St Dev	
1	2.5_3_200	28	0.8721	0.6237	0.8721	7.4117	7.2272	1.3205	19
2	2.5_3_120	30	0.2731	0.1793	0.2958	4.8081	4.7202	0.7411	4
3	2.5_3_60	30	0.1366	0.0836	0.2405	3.5757	3.5727	0.5841	1
4	2.5_2_200	30	0.0859	0.0658	0.0508	7.2019	7.1492	1.0322	0
5	2.5_2_120	30	0.0440	0.0423	0.0228	4.9730	5.0948	0.8381	0
6	2.5_2_60	30	0.0163	0.0133	0.0113	3.3650	3.2885	0.4732	0
7	2.5_1_200	30	0.0029	0.0028	0.0005	7.0038	6.9567	1.1831	0
8	2.5_1_120	30	0.0017	0.0018	0.0004	4.6143	4.5693	1.0416	0
9	2.5_1_60	30	0.0013	0.0014	0.0002	3.4140	3.3807	0.7202	0
10	1.5_3_200	29	0.4781	0.3811	0.4598	6.1251	6.0729	1.2655	13
11	1.5_3_120	30	0.1579	0.1151	0.1974	4.0504	4.1294	0.7928	2
12	1.5_3_60	30	0.0520	0.0430	0.0375	2.8810	2.7231	0.6438	0
13	1.5_2_200	30	0.0604	0.0541	0.0344	6.0405	5.9430	1.3105	0
14	1.5_2_120	30	0.0256	0.0183	0.0199	4.0261	3.9823	0.7026	0
15	1.5_2_60	30	0.0115	0.0095	0.0082	2.6597	2.5377	0.6656	0
16	1.5_1_200	30	0.0023	0.0022	0.0007	5.7868	5.5286	1.5838	0
17	1.5_1_120	30	0.0015	0.0015	0.0002	3.9755	3.6056	1.0820	0
18	1.5_1_60	30	0.0011	0.0012	0.0002	2.8106	2.7171	0.8385	0

Figure 16 and Figure 17 show box plots of the raw data for end-to-end delay and packet loss respectively. The box plots graphically show the general trends discussed for Table 11, where end-to-end delay is greater for scenarios with greater number of streams and VoIP packet size (regardless of mobility), and packet loss is greater for scenarios using larger VoIP packet sizes (regardless of mobility and number of streams). Figure 16 is plotted on a base-ten logarithmic scale to better compare the wide range of responses.

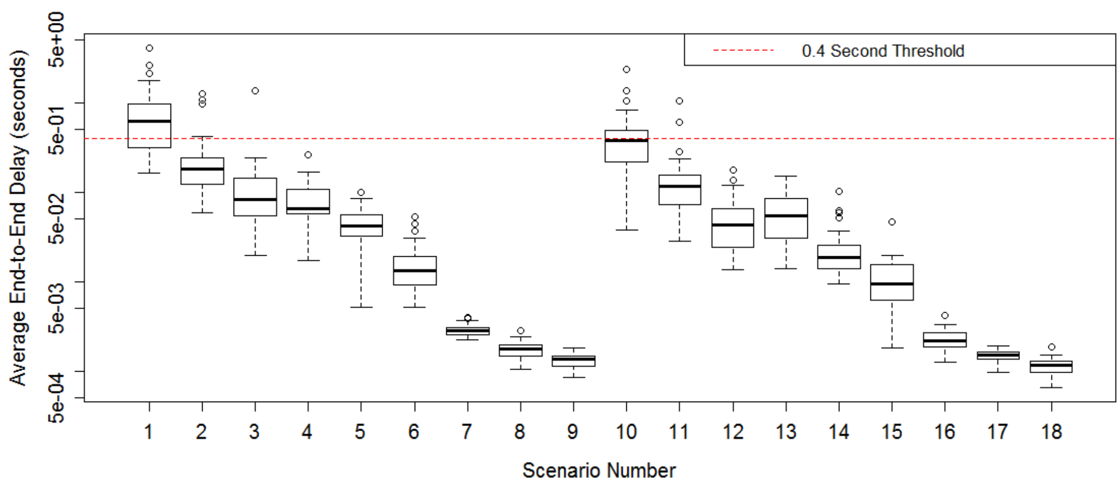


Figure 16: Box Plot of End-to-End Delay Raw Data

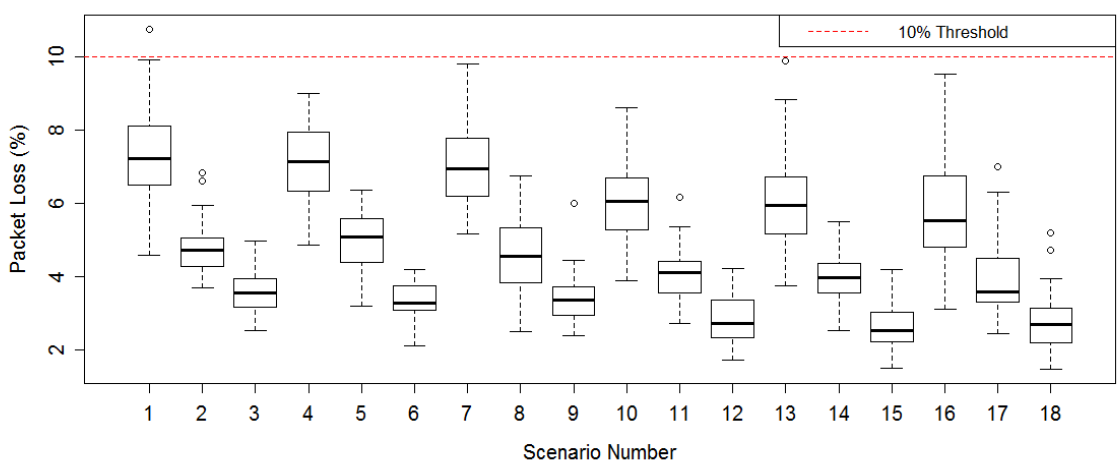


Figure 17: Box Plot of Packet Loss Raw Data

4.2.3 Data Assumption Analysis

This section presents an analysis of the end-to-end delay and packet loss raw data to gain insight into its conformation to normality, homoscedasticity and independence. Valid conclusions about the system can be drawn when these assumptions are at least approximately true. Appendix D contains a complete listing of the data assumption analysis.

The Normal Q-Q plot and Residual Distribution plot, seen in Figure 18 (a) and (c) respectively for Scenario 18, are used to test for normality—where data conforms to a fixed distribution. The Normal Q-Q plot shows conformation to normality when the data that is plotted against a theoretical normal distribution approximately forms a straight line with the solid line and is within the 90% confidence interval (dashed boundaries). Data

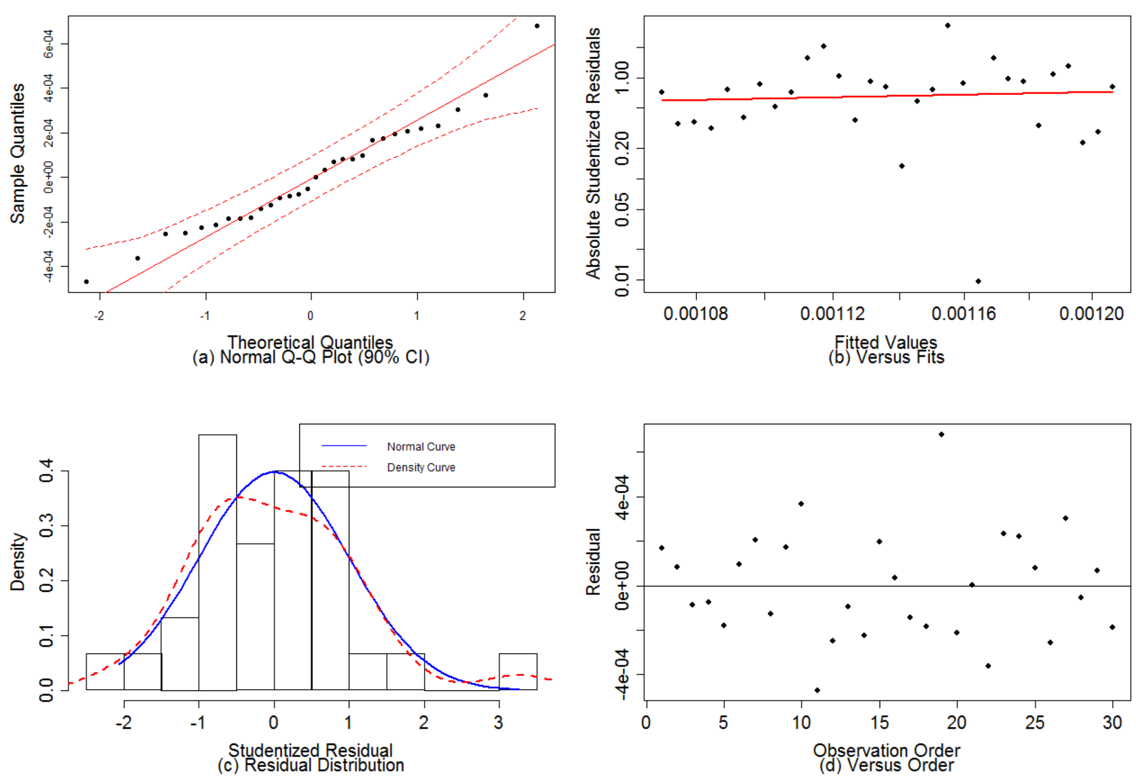


Figure 18: Data Assumption Tests That Conform Well (Scenario 18)

points outside of the 90% confidence interval are considered outliers. The Residual Distribution plot shows conformation to normality when the density curve created by the data set (dashed line) approximately follows a normal distribution (solid line).

Scenario 18 conforms to normality very well; however, Scenario 1 does not conform well to the normality assumptions as seen in Figure 19 (a) and (c). Scenario 1 (three 200 byte streams at jogging speed) is much more stressed than Scenario 18 (one 60 byte stream at walking speed), thus higher variably in performance is experienced. The Normal Q-Q plot for Scenario 1 shows that most of the data points conform to normality, but 21.43% of the data (six outliers of 28 data points retained) does not fall within the 90% confidence interval. Also, the Residual Distribution plot for Scenario 1 shows that the data has a non-normal distribution that is right-skewed with long tails. The skewness

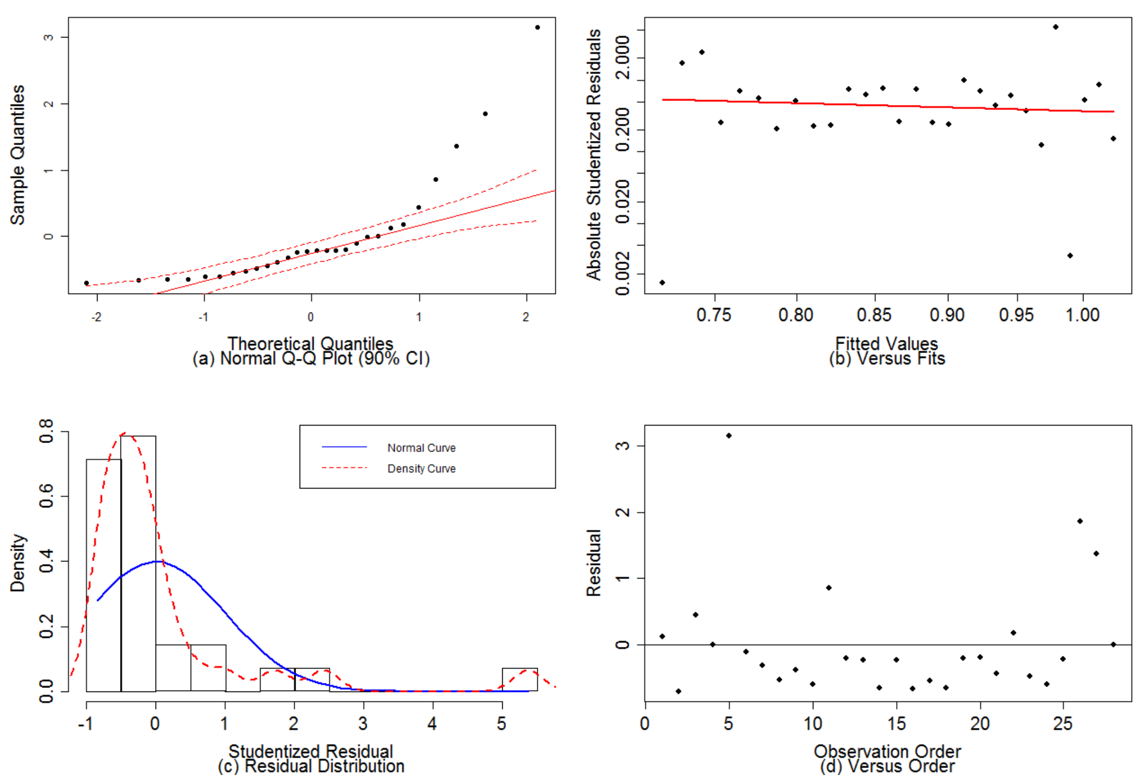


Figure 19: Data Assumption Tests That Do Not Conform Well (Scenario 1)

and long tail of the distribution is due to the observation of outliers in the results. Scenarios 1, 2 and 3 for jogging speeds, and Scenarios 10, 11 and 13 for walking speeds exhibit similar behaviors with respect to the end-to-end delay data. As a result, caution is observed when drawing conclusions from these scenarios when comparing end-to-end delay data. Conversely, all scenarios in this study conform to the assumptions of normality very well when using the packet loss data—as shown in Appendix D.1.

A Versus Fits plot is used to check for homoscedasticity. Simply put, homoscedasticity is the check for constant variance. The Versus Fits plot shows homoscedasticity when the data points form a random horizontal band around an approximately horizontal fitted line that does not form a generalized pattern. Figure 18 (b) and Figure 19 (b) show that homoscedasticity is valid within reason for Scenario 18 and Scenario 1 respectively. The behavior exhibited in all scenarios for end-to-end delay and packet loss data approximate homoscedasticity since residuals about the fitted line are randomized and variance appears constant.

A Versus Order plot is used to test for independence, as seen in Figure 18 (d) and Figure 19 (d). The Versus Order plot shows that the data is independent if no apparent trend is evident. A liner, exponential or sinusoidal trend implies that the data is not random with respect to observation order. All scenarios in this study show random responses with respect to observational order (to include end-to-end delay and packet loss data), thus independence is assumed.

The packet loss data conforms well to the assumptions of normality, homoscedasticity and independence. However, caution must be observed when further analyzing the end-to-end delay data. The most stressed scenarios for jogging speed

(Scenarios 1, 2 and 3) and the most stressed scenarios for walking speed (Scenarios 10, 11 and 12) do not conform well to the normality assumptions. Therefore, after consulting with faculty, it was determined that it is best to isolate comparisons of the most stressed jogging speed scenarios and most stressed walking speed scenarios among each other since these scenarios have an approximately similar non-normal distribution that is right-skewed with long tails. In addition, outliers in this study are retained since the wide variation in response is the true behavior of the system.

4.2.4 Confidence Interval

This study reports the mean of the performance metric with at 90% confidence level. The upper and lower bounds of the confidence interval (CI) are calculated using

$$\left(\bar{x} - z\left(\frac{s}{\sqrt{n}}\right), \bar{x} + z\left(\frac{s}{\sqrt{n}}\right)\right) \tag{1}$$

where is the sample mean, *z* is the quintile of unit variate (1.645 for a 90% confidence interval), *s* is the standard deviation and *n* is the number of samples.

The means of two scenarios are considered statistically similar when the means of the two scenarios fall within each other's CIs. However, if CIs do not contain the mean of the other, then the means may be statistically different for that confidence level. A *t*-test is used to further determine if the means between scenarios are statistically different.

The means and 90% CI's used in the subsequent sections are found in Appendix D.2. A listing of *t*-test results to complement the reported CI's is found in Appendix D.3.

4.3 Impact of VoIP Packet Size

Figure 20 compares end-to-end delay for all scenarios with respect to the VoIP packet size. The three stream scenarios show that voice communication is not sustainable

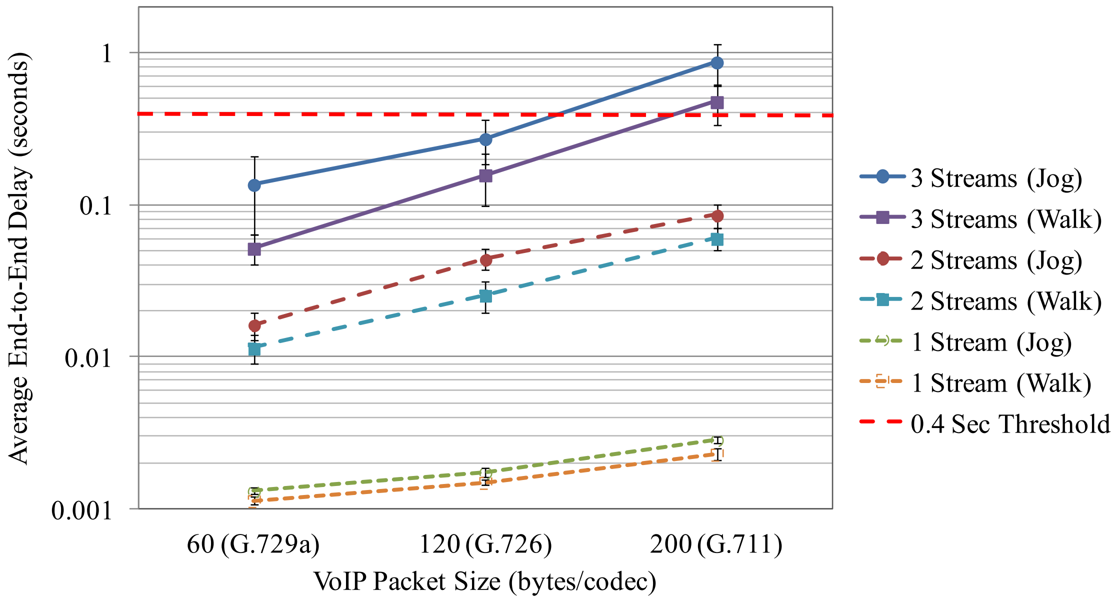


Figure 20: Impact of VoIP Packet Size with 90% CIs (End-to-End Delay)

at walking or jogging speeds when G.711 is employed. An increase of 0.137 seconds is experienced at jogging speeds when G.726 is used instead of G.729a, while an additional increase of 0.599 seconds is observed when G.711 is used in place of G.726—exceeding the end-to-end delay threshold for sustainable VoIP communication. On the other hand, an increase of 0.106 seconds is experienced at walking speeds when G.726 is used instead of G.729a, while an additional increase of 0.320 seconds is observed when G.711 is used in place of G.726—also exceeding the end-to-end delay threshold for sustainable VoIP communication. Although confidence intervals for these scenarios overlap, *t*-tests show that the means are statistically different with the highest *p*-value among comparisons being 0.0548.

Figure 21 presents packet loss results for all scenarios in terms of the VoIP packet size employed. Although none of the scenarios exceed the 10% packet loss threshold, packet loss is significantly worse for scenarios employing G.711 regardless of mobility

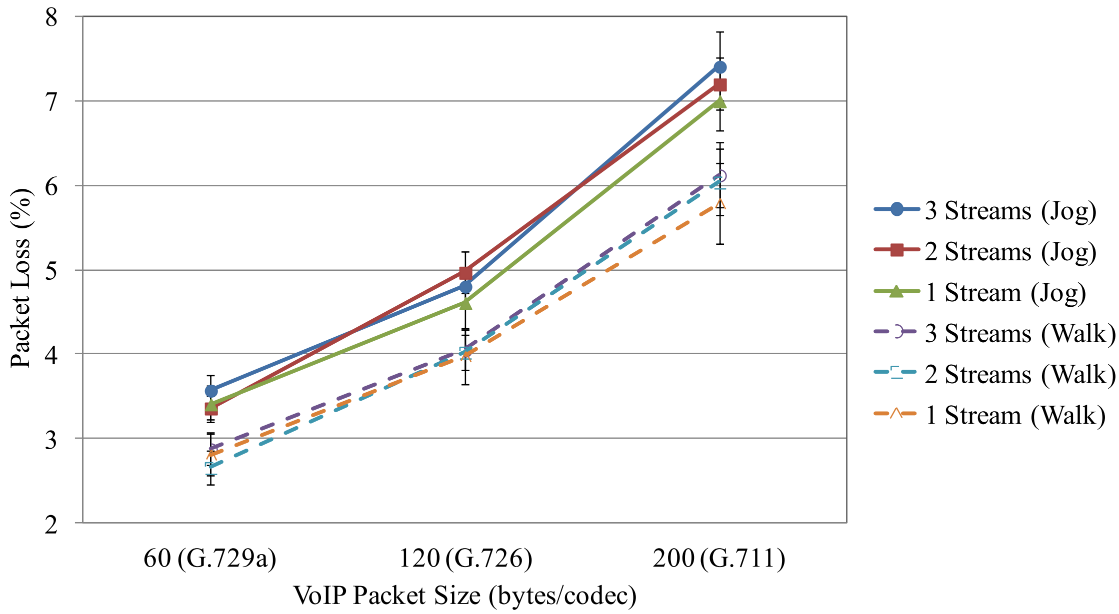


Figure 21: Impact of VoIP Packet Size with 90% CIs (Packet Loss)

and the number of streams used. Specifically, an increase of 1.23 percentage points is experienced at jogging speeds when G.726 with three streams is used instead of G.729a, while an additional increase of 2.60 percentage points is observed when G.711 is used in place of G.726. The one and two stream scenarios for jogging speeds experience similar packet loss behavior. On the other hand, an increase of 1.17 percentage points is experienced at walking speeds when G.726 with three streams is used instead of G.729a, while an additional increase of 2.07 percentage points is observed when G.711 is used in place of G.726. The one and two stream scenarios for walking speeds experience statistically similar packet loss behavior as the three stream scenarios when the codec is varied with the lowest observed *p*-values of 0.1958 and 0.1305 for walking and jogging respectively.

4.4 Impact of the Number of Voice Streams

Figure 22 and Figure 23 compare end-to-end delay for all scenarios with respect to number of streams used for walking and jogging speeds respectively. The G.711 scenarios show that the 0.4 second threshold is exceeded when three streams are used for either jogging or walking speeds. An increase in end-to-end delay of 0.058 seconds is experienced at walking speeds for G.711 when two streams are used rather than one. For jogging speeds, a delay increase of 0.083 seconds is experienced when two streams is used rather than one. However, caution must be observed when comparing the three stream scenarios to the one or two stream scenarios since the three stream scenarios do not conform well to the normal distribution assumption. As a result, the use of three streams for G.711 increases the delay beyond the 0.4 second threshold for both walking and jogging speeds.

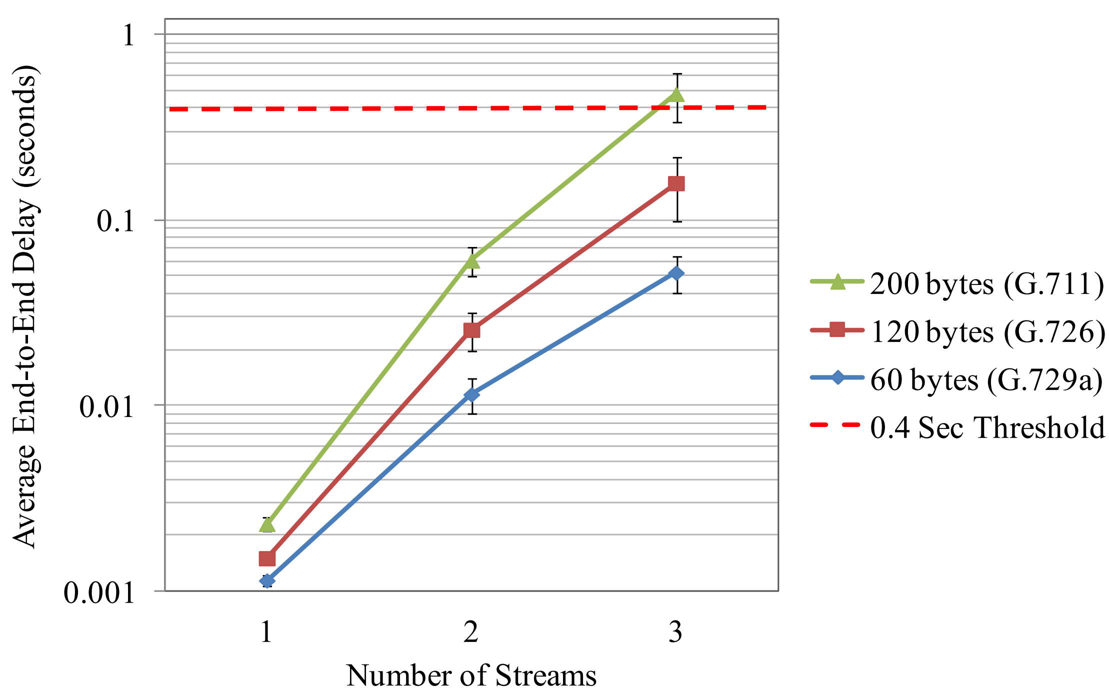


Figure 22: Impact of Number of Streams at Walking Speeds with 90% CIs (End-to-End Delay)

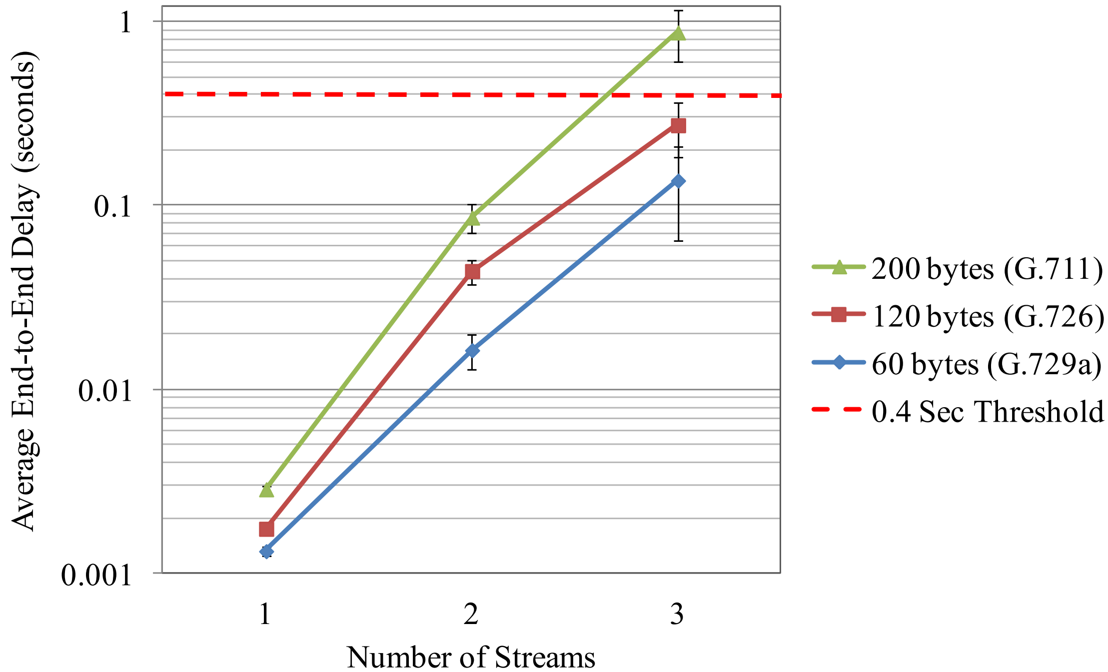


Figure 23: Impact of Number of Streams at Jogging Speeds with 90% CIs (End-to-End Delay)

Figure 24 presents packet loss results for all scenarios in terms of the number of streams used. Recall that all packet loss data conforms well to the assumptions of normality, homoscedasticity and independence; therefore, valid conclusions can be made between scenarios using 1, 2 and 3 streams. Visually, it appears that there is a slight difference in packet loss when the number of streams increases given the use of a particular codec at a given mobility; however, *t*-tests reveal that the means of packet loss are considered statistically similar when the number of streams is varied for any codec and mobility grouping--the lowest *p*-value observed for the packet loss among scenarios with similar codec and mobility is 0.1305.

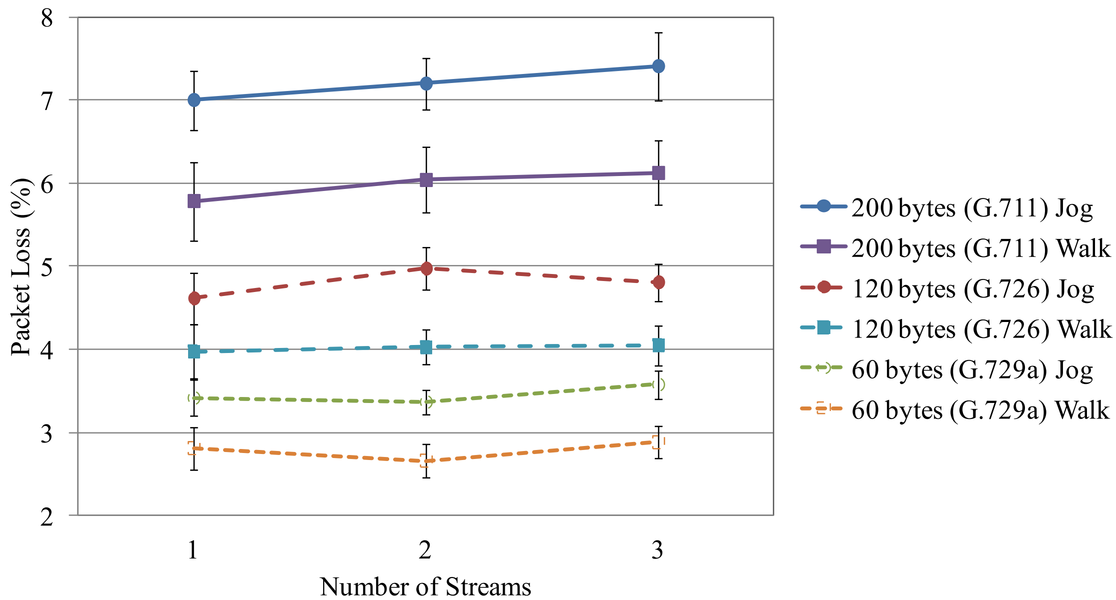


Figure 24: Impact of Number of Streams with 90% CIs (Packet Loss)

4.5 Impact of Mobility

Figure 25 presents end-to-end delay results for all scenarios in terms of mobility. VoIP communication is shown to be unsuitable for three streams of G.711 at either walking or jogging streams since the 0.4 second threshold is exceeded in both scenarios. However, a 0.115 second increase in end-to-end delay is experienced for three streams employing G.726 when mobility is increased from walking to jogging speeds (*p*-value=0.082). Conversely, three streams of G.729a experiences an increase of only 0.085 seconds (*p*-value=0.0663).

Figure 26 shows packet loss results for all scenarios in terms of mobility. An increase of 1.29 percentage points is experienced when the three stream G.711 scenarios increases from walking to jogging speeds. The one and two stream scenarios for G.711 experience similar packet loss behavior. The packet loss means among G.711 scenarios at walking speeds (*p*-value=0.3679) and jogging speeds (*p*-value=0.2218) are statistically

not different. However, the increase in packet loss between walking and jogging speeds are statistically different with the highest observed *p*-value of 0.0079.

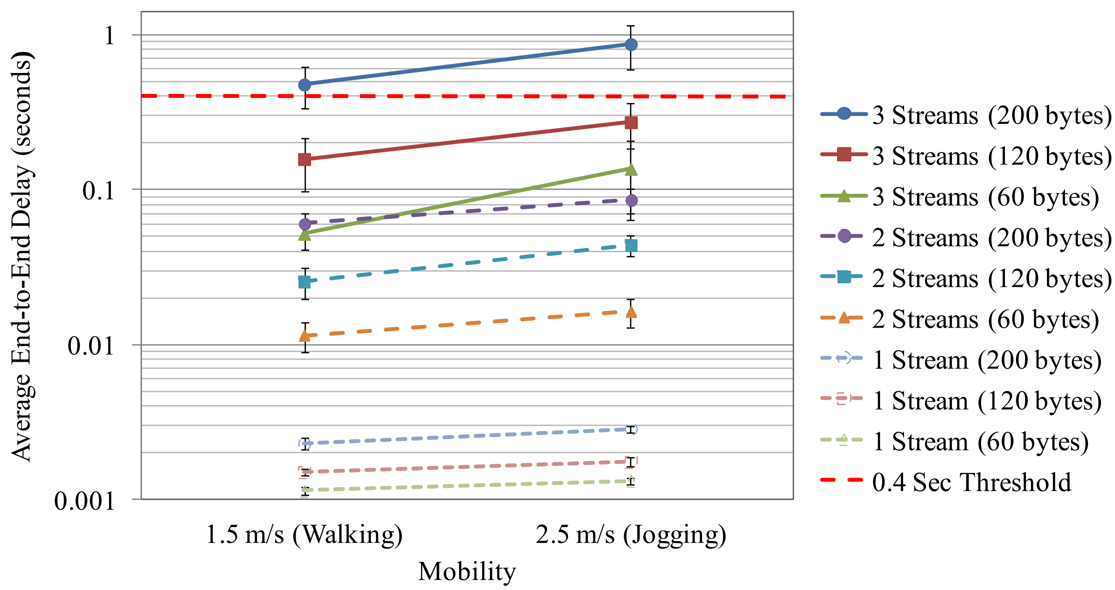


Figure 25: Impact of Mobility with 90% CIs (End-to-End Delay)

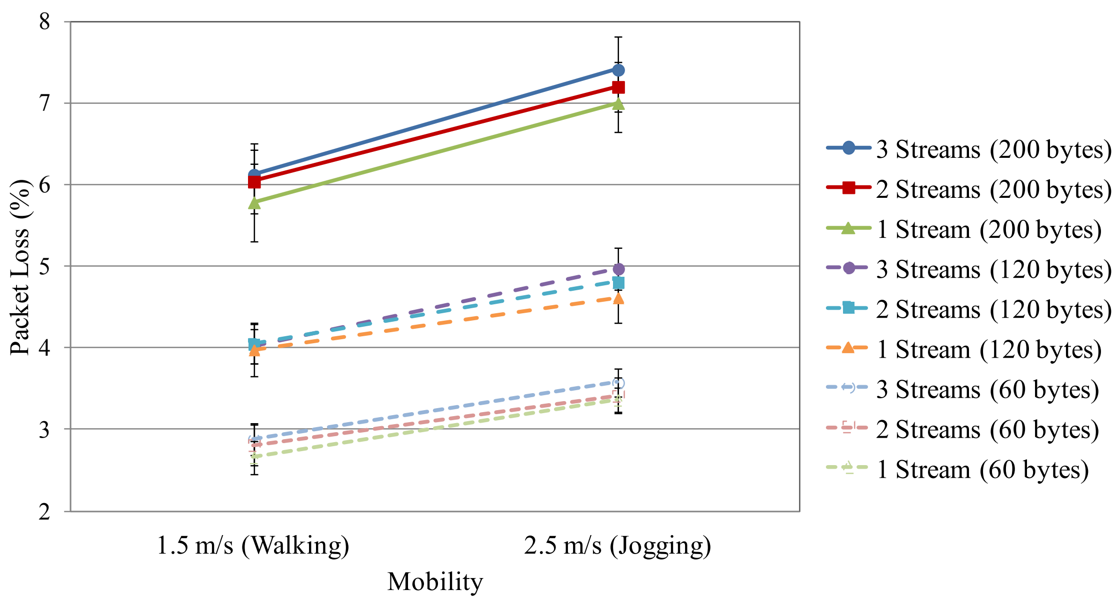


Figure 26: Impact of Mobility with 90% CIs (Packet Loss)

4.6 Analysis of Variance

An Analysis of Variance (ANOVA) is used to determine which factor submitted to the system has the most significant impact on the response of the system with respect to end-to-end delay and packet loss.

The ANOVA results include source, degrees of freedom (DF), sum of squares (SS), mean square (MS), F-Value, and the chance of obtaining a greater F-Value—Pr(>F). The source is a factor in which variation is observed. There is one source for each factor (i.e., Mobility, Streams, and Packet Size). Residuals are those variances that are not accounted for by the primary factors. The DF describes the statistical value that is free to vary. The SS describes the deviation observed within each source of deviation. The MS is calculated by dividing the SS by its associated DF. The F-Value is calculated by dividing the MS of the source by the MS of the residual. Higher F-Values indicate that a particular source produces a significant model effect. The Pr(>F) describes the probability (*p*-value) of obtaining a greater F-Value.

The ANOVA results for end-to-end delay and packet loss are shown in Table 12. The ANOVA results for end-to-end delay shows that the highest F-Value observed is 12.3376 for "Streams" with a *p*-value of 0.003449. This demonstrates that the number of streams submitted to the system has the greatest impact on the end-to-end delay observed. Additionally, the highest observed F-Value for packet loss is 943.705 for "Packet Size"

Table 12: ANOVA Results

	End-to-End Delay					**Packet Loss**				
Source	DF	SS	MS	F-Value	Pr(>F)	DF	SS	MS	F-Value	Pr(>F)
Mobility	1	0.02302	0.02302	0.8882	0.361955	1	3.564	3.564	91.1266	1.66E-07
Streams	1	0.3198	0.3198	12.3376	0.003449	1	0.129	0.129	3.3038	0.090558
Packet Size	1	0.14353	0.14353	5.5371	0.033763	1	36.914	36.914	943.705	3.01E-14
Residuals	14	0.36289	0.02592			14	0.548	0.039		

with a *p*-value of 3.01E-14. This indicates that the size of the VoIP packet submitted to the system has the greatest impact on the packet loss observed.

4.7 Interpretive Analysis

The previous sections provide an observational analysis of the impact of varying the factor levels on the performance metrics. However, an interpretive analysis is needed in order to give insight as to why an increase in end-to-end delay and packet loss is experienced when the factor levels are increased.

It is intuitive that VoIP packets with larger payloads will have inherently greater end-to-end delays because larger packets have longer transmission delays; however, other dynamics also contribute to the increased delay. These other factors include lack of MPR optimization and contention for the wireless medium.

Recall that OLSR attempts to reduce flooding with the use of MPRs. Flooding is optimized when the MPR set for a node is reduced to the minimum nodes needed to maintain reach of the entire two-hop neighbor set. Increased mobility forces the network topology to change faster than the optimal MPR set can be calculated. This results in nodes having a larger MPR set that subsequently results in higher routing traffic overhead which contributes to the end-to-end delay experienced by the VoIP packets.

The nodes in the MANET are continually competing for the wireless medium, therefore the use of buffers is required until the medium becomes available. Buffered VoIP packets are not only subject to the transmission delay between nodes, but are also subject to the transmission delays of packets already in the queue. The effect of end-to-end delay is much more pronounced for VoIP packets with higher payloads and scenarios with a higher number of streams.

Packet loss in the MANET is observed at the IP and MAC layers. A node drops packets at the IP-layer if the route table look-up fails to yield a route toward the destination. OLSR is a proactive protocol and strives to have routes readily available; however, periodic update may quickly become invalid due to the dynamic nature of MANETs. This may be due to a node that was once within transmission range now being outside of transmission range. Packets can also be dropped at the MAC layer if a node is unable to transmit a packet before retry threshold is exceeded. The retry threshold is exceeded if other nodes are occupying the medium at the same time that another node is trying to transmit (or retransmit) a data frame within a reasonable amount of time but is unable to do so. Recall that the hidden node and exposed node problem is well documented in MANETs [CDL05] [Gas05] [TYH06]. The retry threshold is also exceeded if an ACK is never received for a transmitted packet within a reasonable amount of time. Even though VoIP packets operate using UDP and ACKs are not needed at the IP layer, the MAC layer for 802.11 requires the atomic process of receiving ACKs for all data frames transmitted by a node [Gas05].

4.8 Summary

This chapter presents and analyzes 1074 data points collected over 18 scenarios. The exploratory data analysis shows that scenarios involving three streams experience high variability in performance metrics. In turn, caution is used in observing Scenarios 1, 2, 3, 10, 11, and 12 since these scenarios do not conform well to the assumptions of normality. These scenarios are only compared amongst themselves and not with the normally distributed data.

This study reveals that VoIP communication is not sustainable at walking (1.5 m/s) or jogging (2.5 m/s) when three simultaneous streams are used with each utilizing G.711 (resulting in a 200 byte VoIP packet). Additionally, G.729a (resulting in a 60 byte VoIP packet) experiences the least amount of end-to-end delay and packet loss of the three codecs compared and is best suited for MANETs. Specifically for the three stream scenarios, G.729a performed 0.736 seconds faster at jogging speed and 0.426 seconds faster at walking speed then the worst performing codec.

V. Conclusions

The data collected during simulation is examined to determine the performance impact of VoIP codecs in various MANET environments using OLSR. The hypothesis of this research is supported and shows that VoIP codecs with smaller data payloads outperform VoIP codecs that carry larger data payloads as the MANET becomes stressed with various workloads. Additionally, a failure point is reached with some of the workloads submitted to the system under test.

Varying combinations of workloads are submitted to the simulated MANET to determine voice performance in a stressed environment. Performance metrics are compared against established benchmarks to determine if thresholds for unacceptable voice quality are exceeded. The thresholds for acceptable VoIP communication are an end-to-end delay less than 400 msec and packet loss not exceeding 10%.

5.1 Conclusions

Performance analysis reveals that VoIP communication using G.711 is not sustainable at walking or jogging speeds when three simultaneous streams are used since the end-to-end delay threshold is exceeded. Additionally, G.729a is best suited for MANETs since it outperforms the other codecs used in terms of end-to-end delay and packet loss. Specifically for the three stream scenarios, G.729a performed 0.736 seconds faster at jogging speed and 0.426 seconds faster at walking speed than the worst performing codec. Packet loss in all experiments observed does not exceed the 10% threshold.

5.2 Future Work

The failure point of voice communications can be studied further by using different routing protocols. This can be accomplished by comparing the OLSR results with those of other proactive or reactive protocols by using the simulation environment already created. This type of work can compare the effects of routing protocols between each other on multimedia traffic in terms of communication thresholds.

MANETs can be studied in a vehicle setting. Vehicular ad hoc networks (VANETs) are used for communications between vehicles and roadside equipment. IEEE 802.11p is an active standard for wireless access in vehicular environments (WAVE) [IEE10].

Unmanned aerial vehicle (UAV) swarms is another technology that can benefit from wireless routing protocol research. Research on UAV swarms has been conducted using simulations [Lid08]. However, a live experimental test bed of a UAV swarm providing a video feed over multiple hops can determine the performance benefits of one routing protocol over another.

5.3 Relevance of Work

MANETs are a growing technology that is applicable to military and commercial applications. The use of MANETs in a combat environment allow for reliable voice or data communication without the need for infrastructure (a single point of failure). The commercial sector can benefit from the use of MANETs by providing customers Internet access in non-conventional ways. The continued maturity of MANETs will allow seamless integration for infrastructure based access points to provide Internet capabilities to a network of ad hoc nodes.

5.4 Summary

This chapter presents conclusions from the research. The research significance and recommendations for future work are also discussed.

Appendix A. OPNET Simulation Setup

This appendix explains the necessary steps needed in order to create and run these simulations in OPNET. Section A.1 gives an overview on scenario creation and setup. Section A.2 gives details on how VoIP packets are added to the MANET. Section A.3 provides how mobility is adjusted for the ad hoc nodes. Section A.4 lists the average run time for each scenario.

A.1 Scenario Creation and Setup

The initial project is created using the startup wizard in OPNET. Table 13 outlines the initial setup values used to create the simulation space.

Table 13: Initial Scenario Setup Parameters

Parameter	Setting
Initial Topology	Create empty scenario
Network Scale	Campus
Size	1000 x 1000 meters
Model Family	MANET

Ad hoc nodes are generated using the Wireless Network Deployment suite found in the topology menu in OPNET. Table 14 outlines the parameters used to deploy the MANET. Note that an initial speed of 2.5 m/s is used for mobility to emulate jogging speeds. The mobility model used by the ad hoc nodes is later modified to a speed of 1.5 m/s to emulate walking speeds—see Section A.3.

Figure 27 shows the random placement of the ad hoc nodes within the simulation area while Table 15 lists the initial coordinates of the nodes.

Table 14: Initial Wireless Deployment

Parameter	Setting
Location	New Subnet
Coordinates (X, Y)	(0.00, 0.00)
Technology	WLAN (Ad-hoc)
Operational Mode	802.11g
Data Rate	54 Mbps
Ad-hoc Routing Protocol	OLSR
Geographical Overlay	None
Area	1000000 meters^2
Node Placement	Random
Node Model	manet_station
Count	30 Nodes
Trajectory Information	Random Waypoint (Auto Create)
Speed	2.5 m/s
Area of Movement	Within Network
Altitude	0.00 meters

Table 15: Initial Placement of 30 Node MANET (Used In All Experiments)

Node	X	Y	Node	X	Y	Node	X	Y
1	-307.0087	-309.8109	**11**	-146.0883	150.2936	**21**	288.1284	-302.9942
2	-210.1458	247.7381	**12**	395.2746	-338.5665	**22**	363.6073	335.8809
3	420.3605	-230.3793	**13**	180.5436	-86.6361	**23**	-24.9792	-319.5517
4	-185.1335	-29.6736	**14**	-343.0123	328.6511	**24**	-389.2893	10.3452
5	-337.8882	-391.9985	**15**	-125.4690	-314.9251	**25**	436.7043	305.2206
6	-168.9451	-352.3949	**16**	444.4156	67.5224	**26**	426.6108	-159.3840
7	-452.2475	-221.9849	**17**	-124.7360	-215.7303	**27**	5.5141	371.8854
8	-233.1590	-275.1702	**18**	-134.7395	-154.3755	**28**	2.0495	-263.9422
9	-464.6757	256.2560	**19**	53.8904	180.1269	**29**	-164.7507	159.0372
10	194.5098	-83.7343	**20**	315.9509	216.0022	**30**	-385.2912	209.7804

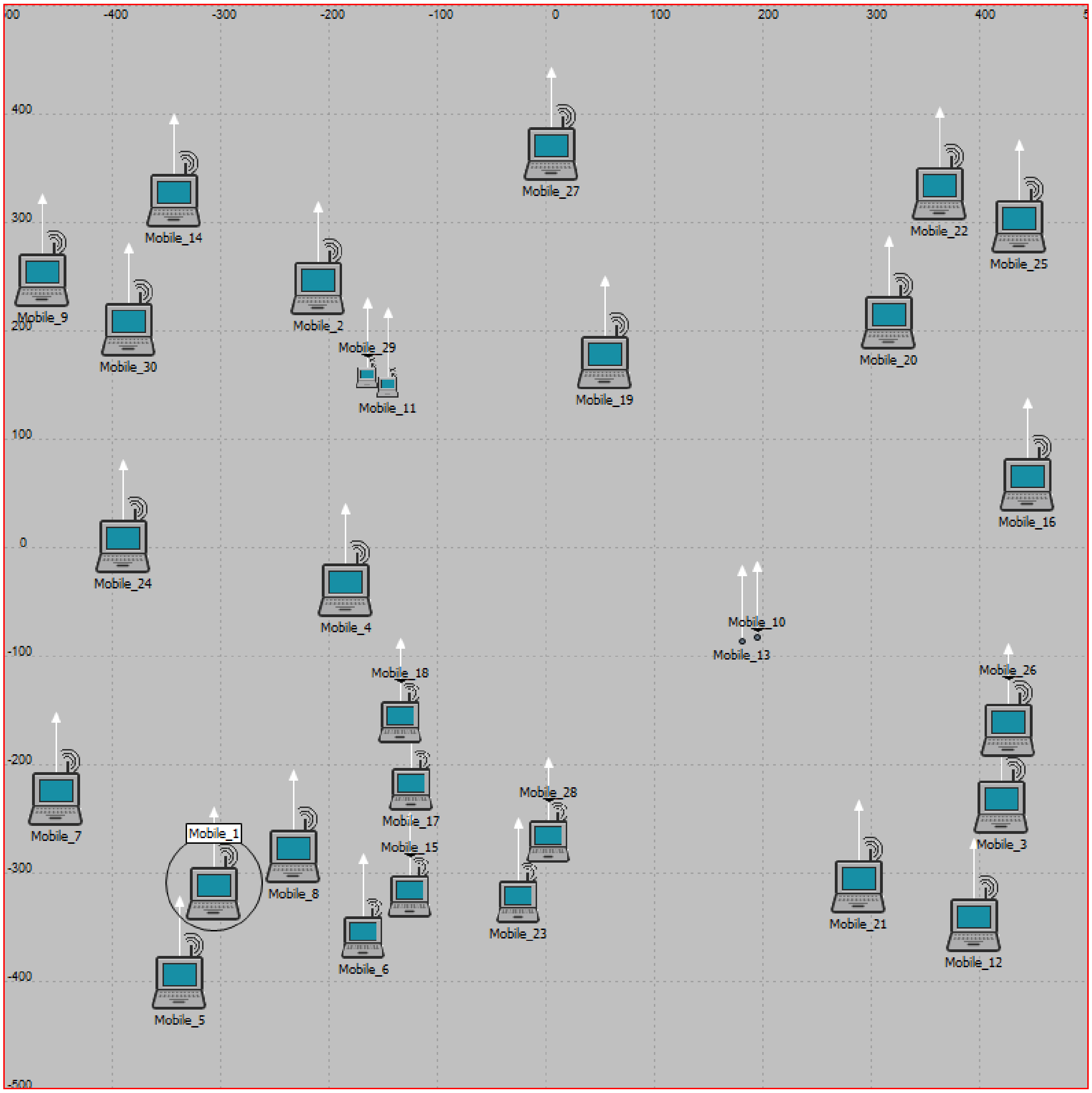
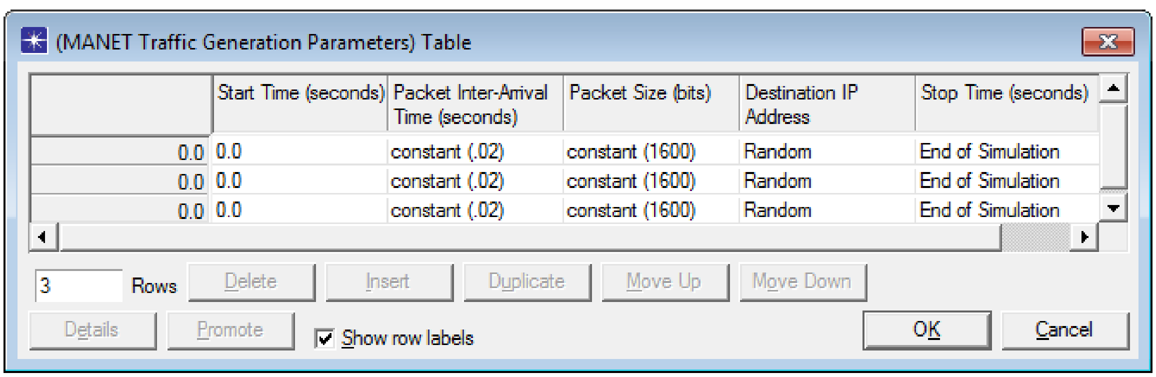


Figure 27: Initial Placement of 30 Node MANET

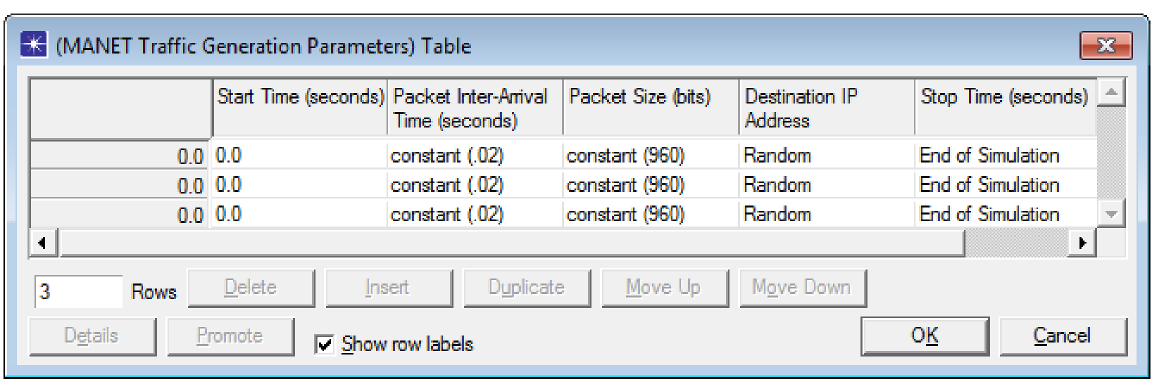
A.2 VoIP Packet Generation

All VoIP traffic is generated from the source node (Mobile_1) and sent to a random destination within the network. Each stream has its own randomly assigned destination that remains the same for the duration of the simulation. The three codecs emulated in this study have the same traffic generation rate of 50 packets per second which results in a constant packet inter-arrival time of 0.02 seconds (1/50 packets per second). Accounting for 40 bytes of RTP/UDP/IP header and audio payload, VoIP packets have a size of 200, 120 and 60 bytes for G.711, G.726 and G.729a respectively. This further translates to 1600, 960 and 480 bits for G.711, G.726 and G.729a respectively. Figure 28 shows VoIP packet generation for G.711. Figure 29 shows VoIP packet generation for G.726. Figure 30 shows VoIP packet generation for G.729a.



	Start Time (seconds)	Packet Inter-Arrival Time (seconds)	Packet Size (bits)	Destination IP Address	Stop Time (seconds)
0.0	0.0	constant (.02)	constant (1600)	Random	End of Simulation
0.0	0.0	constant (.02)	constant (1600)	Random	End of Simulation
0.0	0.0	constant (.02)	constant (1600)	Random	End of Simulation

Figure 28: G.711 VoIP Packet Generation



	Start Time (seconds)	Packet Inter-Arrival Time (seconds)	Packet Size (bits)	Destination IP Address	Stop Time (seconds)
0.0	0.0	constant (.02)	constant (960)	Random	End of Simulation
0.0	0.0	constant (.02)	constant (960)	Random	End of Simulation
0.0	0.0	constant (.02)	constant (960)	Random	End of Simulation

Figure 29: G.726 VoIP Packet Generation

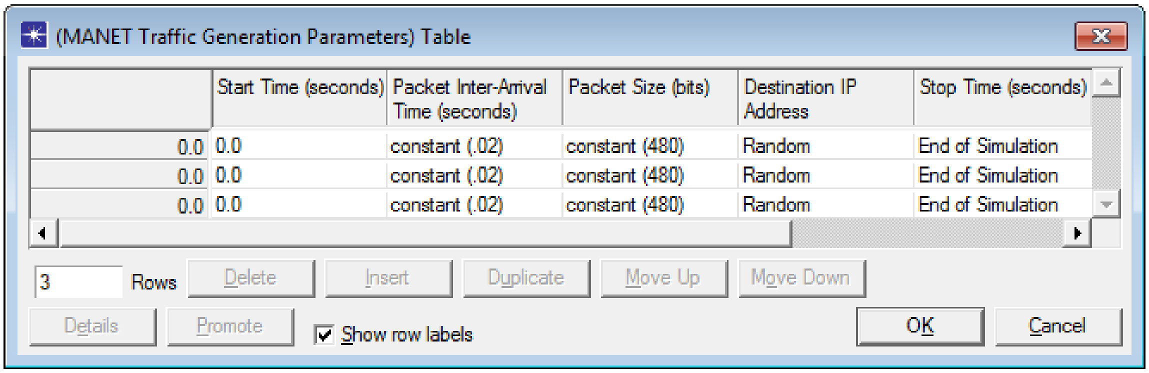


Figure 30: G.729a VoIP Packet Generation

A.3 Mobility

The mobility model used for this study is the RWM. The mobility model is adjusted slightly to reflect walking speeds and jogging speeds. These attributes are adjusted in the Mobility Configuration Profile located in the Top Parent Directory. Figure 31 shows a walking speed of 1.5 m/s, while Figure 32 shows a jogging speed of 2.5 m/s. The RWM operates within the simulation area.

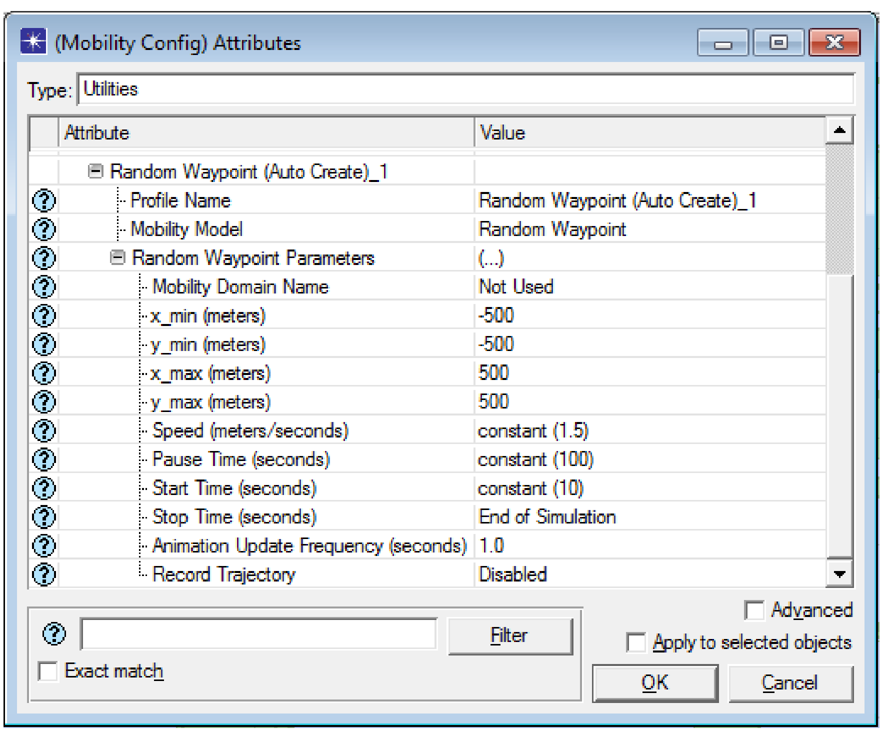


Figure 31: RWM Attributes for Walking Speeds

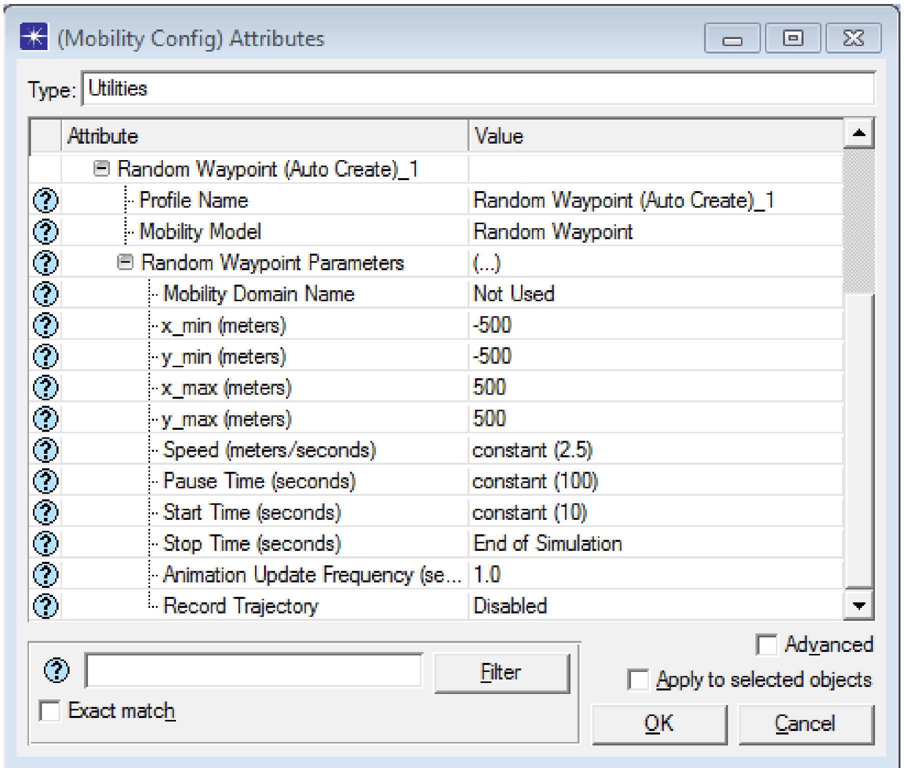


Figure 32: RWM Attributes for Jogging Speeds

A.4 Average Simulation Run Times

Each scenario is repeated thirty times with different seeds for each repetition. Table 16 shows the average time it takes to run two hours of simulation time for one repetition given the scenario.

Table 16: Average Run Times for Each Scenario

Scenario (Speed_Streams_Size)	Average Run Time (Minutes:Seconds)	Scenario (Speed_Streams_Size)	Average Run Time (Minutes:Seconds)
3_4_200	19:53	**1dot5_4_200**	20:23
3_4_120	20:26	**1dot5_4_120**	21:45
3_4_60	21:34	**1dot5_4_60**	20:01
3_3_200	16:57	**1dot5_3_200**	17:36
3_3_120	17:51	**1dot5_3_120**	16:28
3_3_60	16:54	**1dot5_3_60**	17:01
3_1_200	8:57	**1dot5_1_200**	8:12
3_1_120	8:26	**1dot5_1_120**	8:16
3_1_60	8:38	**1dot5_1_60**	8:10

Appendix B. Validation

This appendix explains the validation process taken to ensure the model functions as intended. Section B.1 explains Wireless Link Validation of 802.11g. Section B.2 explains routing validation in a generic sense. Section B.3 describes how OLSR is validated. Section B.4 details RWM validation.

B.1 Wireless Link Validation

The IEEE 802.11g wireless link is validated by observing the amount of packet loss experienced by varying the distances between the sender node (Mobile_1) and receiver nodes (Mobile_2) incrementally from 300 to 450 meters. Results are listed in Table 8. Figure 33 shows Mobile_1 and Mobile_2 at a distance of 375 meters apart.

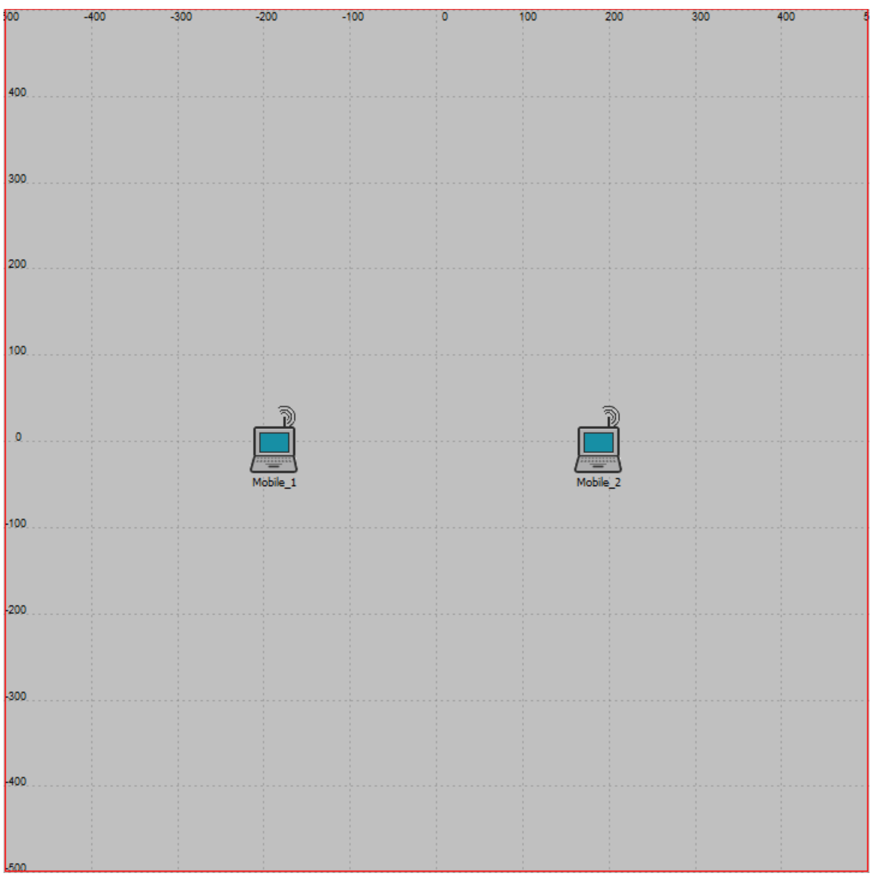


Figure 33: Wireless Link Validation at 375 Meters Apart

B.2 Routing Validation

Mobile_1, Mobile_3 and Mobile_5 are outside of each other's broadcast range with a distance of approximately 566 meters between them. However, Mobile_2 is within range of Mobile_1 and Mobile_3 with a distance of approximately 283 meters between nodes. Mobile_4 is also within broadcast range at a distance of 283 meters, but with Mobile_3 and Mobile_5. All nodes are stationary. Mobile_1 sends 86300 packets, and 86300 packets are received by Mobile_5 with a total of 4 hops between nodes for all packets sent.

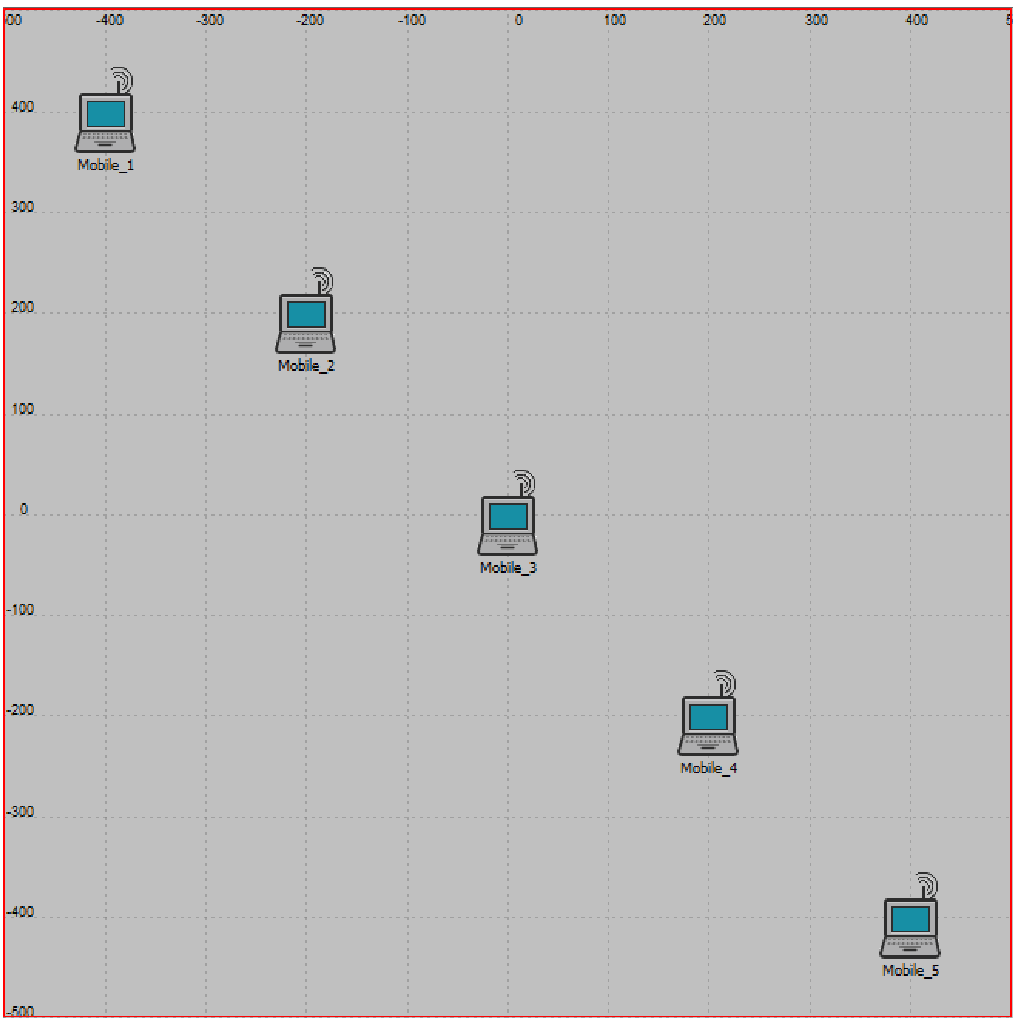


Figure 34: Routing Validation

B.3 OLSR Validation

The source (Mobile_1) and destination (Mobile_3) nodes are stationary and are outside of each other's broadcast range with a distance of 600 meters between them. The intermediate node (Mobile_2) is initially outside of the source and destination nodes broadcast range, but then moves within range of both nodes at a speed of 2.5 m/s.

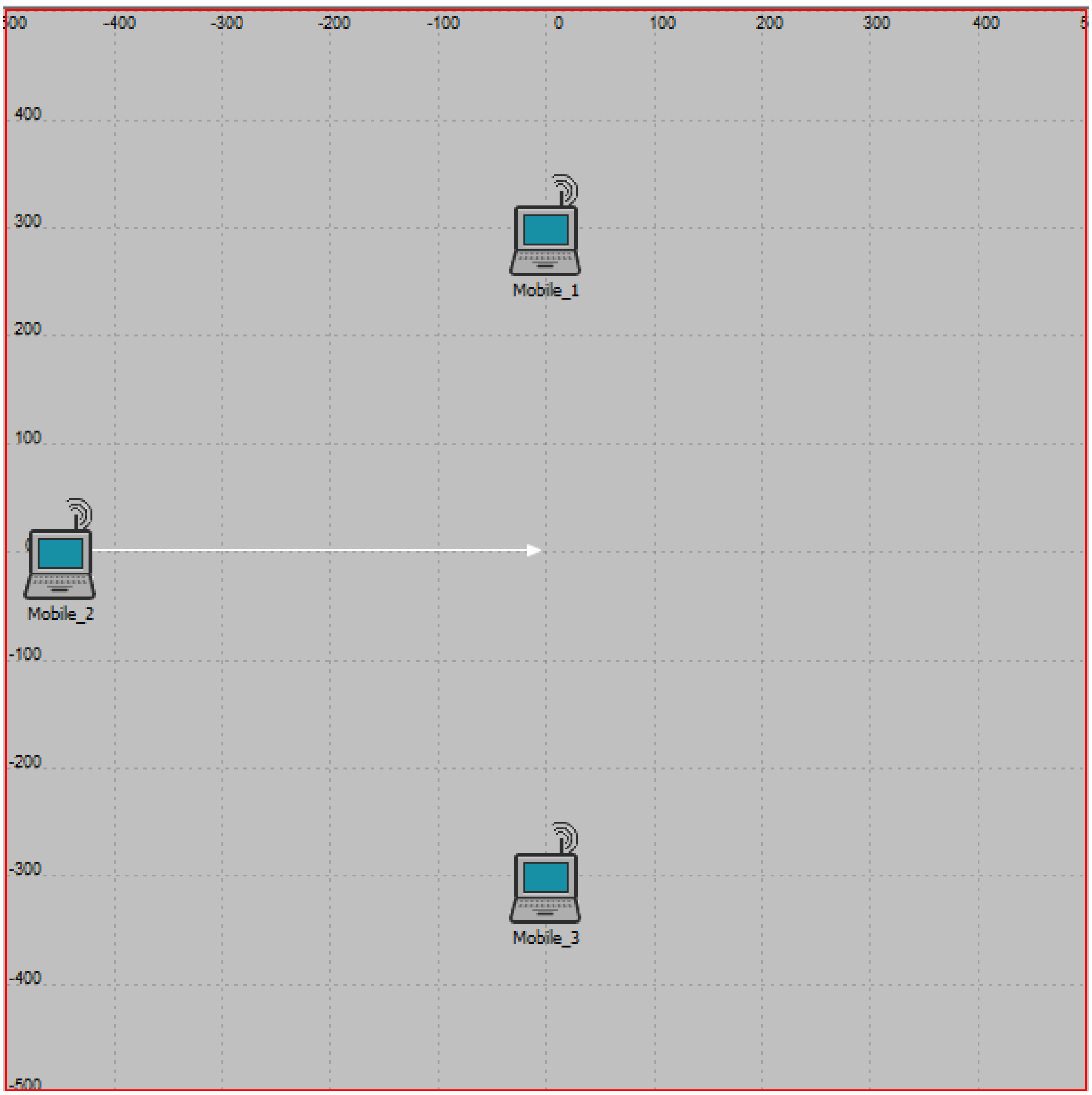


Figure 35: OLSR Validation

The Mobile_2 node exchanges routing information with the Mobile_1 and Mobile_3 nodes when within range. The Mobile_1 node then selects the Mobile_2 node as a MPR and forwards packets to the destination. Figure 36 shows the routing traffic received between the nodes and the selection of Mobile_2 as the MPR. When the MPR status is 0 then Mobile_2 is not selected; however, when the MPR status is 1 then Mobile_2 is selected. The OLSR routing protocol behaves as expected.

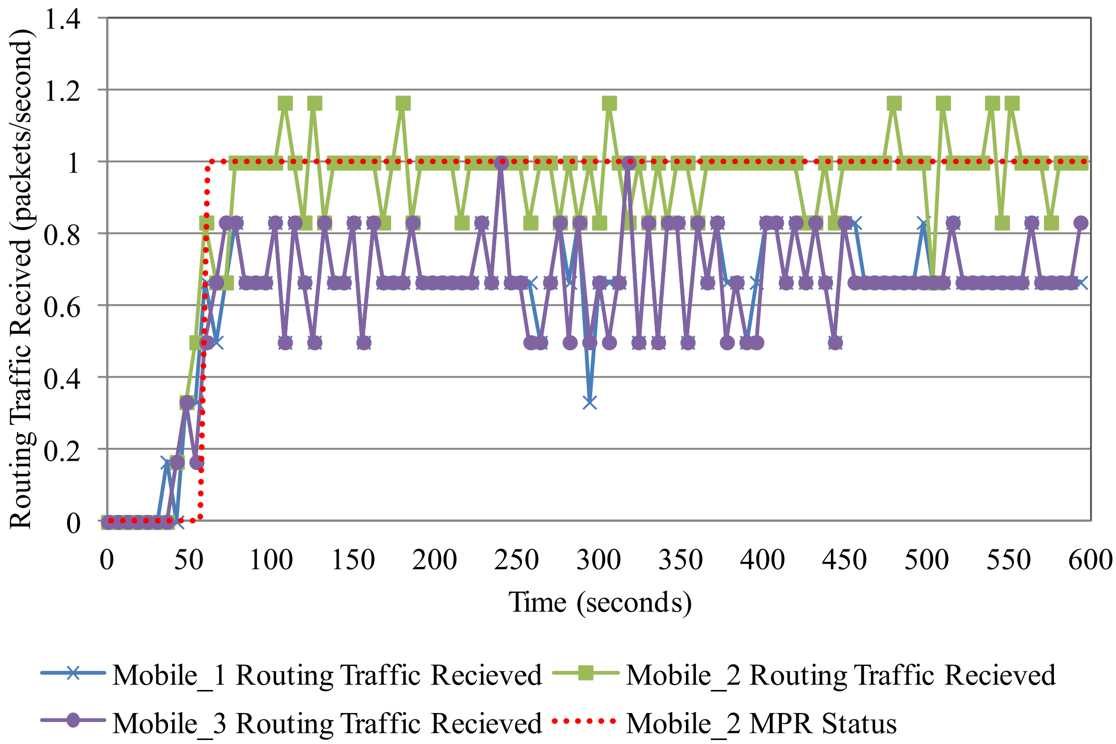


Figure 36: OLSR Routing and MPR Status Validation

Figure 37 further validates that the OLSR routing protocol is functioning as intended. Mobile_1 is sending VoIP traffic at a rate of 50 PPS. Traffic is routed to Mobile_3 after Mobile_2 is selected as the MPR as a result of being within the transmission range of both nodes.

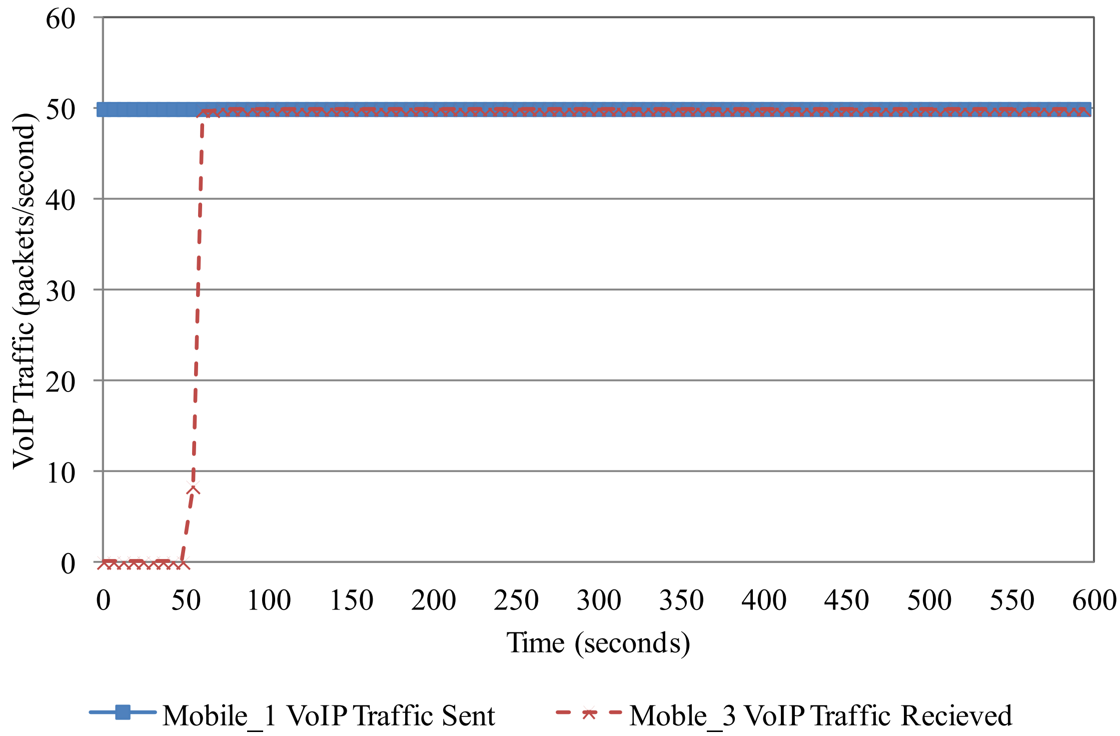


Figure 37: OLSR Validation with VoIP Traffic

B.4 RWM Validation

The mobility model is validated by observation. Ad hoc nodes exhibit movement pattern consistent with the RWM. Figure 38 shows that each node randomly selects a destination and travels to the destination at a fixed speed. Once the destination is reached, the node pauses for 100 seconds, selects a new destination and then travels to the new destination at the same fixed speed. All nodes stay within the bounds of the 1,000 m by 1,000 m simulation area.

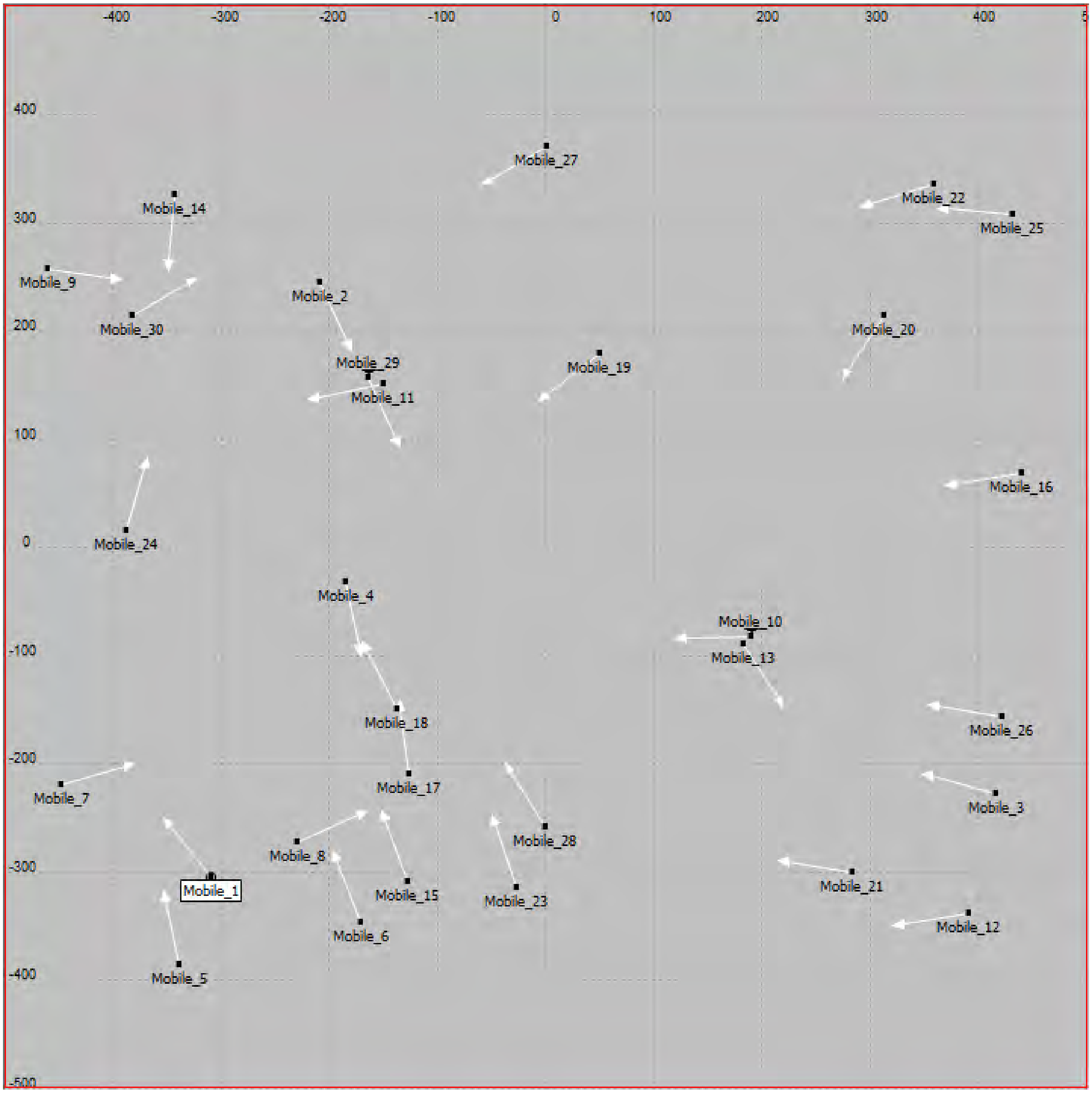


Figure 38: RWM Validation

Appendix C. Raw Data Collection

This section presents all the raw data collected for each repetition for all scenarios. The naming convention for each scenario uses the following format: "Speed" _"Number of Streams"_ "VoIP Packet Size".

C.1 End-To-End Delay

Table 17, Table 18 and Table 19 list the end-to-end delay (in seconds) experienced in the MANET for all scenarios and each repetition.

Table 17: Average End-to-End Delay Raw Data (1 of 3)

Scenario	Seed									
	149	953	811	1019	37	17	397	683	277	977
1	1.14949	0.30711	1.44174	0.98727	4.13210	0.86084	0.63724	0.41841	29.4943	0.54584
2	0.37421	0.07246	0.13053	0.17409	0.23914	0.18448	0.11180	0.10651	0.16216	0.21581
3	0.03319	0.16008	0.20454	0.08984	0.24454	0.02965	0.18929	0.05616	0.06570	0.14305
4	0.10813	0.26025	0.08033	0.06262	0.16639	0.02671	0.05592	0.13338	0.02859	0.13704
5	0.04236	0.04895	0.04994	0.01741	0.07632	0.02138	0.08565	0.03194	0.00508	0.03265
6	0.01466	0.00751	0.01057	0.03030	0.02246	0.05300	0.00996	0.00841	0.01437	0.00782
7	0.00365	0.00281	0.00295	0.00394	0.00284	0.00283	0.00264	0.00290	0.00242	0.00223
8	0.00142	0.00122	0.00177	0.00105	0.00180	0.00159	0.00194	0.00139	0.00183	0.00218
9	0.00180	0.00139	0.00147	0.00108	0.00110	0.00139	0.00097	0.00122	0.00140	0.00136
10	0.12649	0.17606	0.15630	0.48070	0.39229	2.33283	0.82256	0.33127	0.44182	0.40234
11	0.03345	0.10516	0.12206	0.07174	0.15744	0.60313	0.05762	0.02901	0.05540	0.10323
12	0.01446	0.04387	0.17435	0.03513	0.13461	0.09864	0.06221	0.06877	0.03247	0.03838
13	0.06844	0.08397	0.05760	0.11000	0.02687	0.01792	0.02432	0.03667	0.05231	0.02116
14	0.03026	0.01371	0.01872	0.01538	0.00943	0.02448	0.01269	0.06128	0.05213	0.10208
15	0.00574	0.01913	0.01607	0.00904	0.00749	0.00811	0.00483	0.01210	0.00613	0.01409
16	0.00264	0.00124	0.00322	0.00250	0.00128	0.00214	0.00283	0.00204	0.00187	0.00190
17	0.00122	0.00154	0.00150	0.00150	0.00183	0.00097	0.00158	0.00178	0.00118	0.00124
18	0.00124	0.00116	0.00099	0.00101	0.00091	0.00119	0.00130	0.00098	0.00128	0.00148

Table 18: Average End-to-End Delay Raw Data (2 of 3)

Scenario	Seed									
	2833	**929**	**107**	**421**	**463**	**1511**	**809**	**449**	**1109**	**601**
1	0.31950	1.76997	0.68893	0.66199	0.22742	0.63378	0.19184	0.29492	0.18129	0.61364
2	0.17413	1.26310	0.95881	0.12614	0.24120	0.12683	0.19351	0.14570	0.30535	0.25899
3	0.05940	0.11279	0.05436	0.11753	0.05852	0.08585	1.37283	0.07038	0.03211	0.01934
4	0.06525	0.09900	0.01727	0.06143	0.09703	0.04443	0.05640	0.09229	0.06370	0.07264
5	0.02036	0.04254	0.03306	0.08123	0.04232	0.04904	0.04821	0.03411	0.04177	0.02313
6	0.01088	0.01876	0.01529	0.00514	0.01995	0.01908	0.01957	0.01560	0.01193	0.00905
7	0.00281	0.00231	0.00220	0.00264	0.00388	0.00301	0.00253	0.00303	0.00309	0.00318
8	0.00240	0.00206	0.00180	0.00185	0.00217	0.00142	0.00279	0.00160	0.00139	0.00204
9	0.00084	0.00107	0.00153	0.00131	0.00145	0.00099	0.00147	0.00148	0.00143	0.00179
10	0.14098	0.38110	0.14602	0.21492	0.16495	0.46558	0.25563	0.57596	1.05783	0.22588
11	0.16176	0.15520	0.02789	0.04526	1.03493	0.28096	0.13004	0.11800	0.11591	0.18792
12	0.04761	0.04636	0.02268	0.11966	0.02240	0.05400	0.06635	0.01658	0.03396	0.03759
13	0.15139	0.06937	0.04657	0.03050	0.01389	0.09063	0.03042	0.02882	0.09298	0.03933
14	0.01493	0.02564	0.02413	0.02302	0.01798	0.01562	0.01717	0.03335	0.02178	0.00964
15	0.01531	0.01330	0.00182	0.00426	0.00855	0.00555	0.01941	0.01031	0.01573	0.01825
16	0.00226	0.00415	0.00218	0.00240	0.00332	0.00156	0.00195	0.00174	0.00201	0.00319
17	0.00164	0.00150	0.00185	0.00151	0.00149	0.00129	0.00141	0.00142	0.00156	0.00189
18	0.00065	0.00087	0.00103	0.00091	0.00133	0.00117	0.00100	0.00096	0.00183	0.00095

Table 19: Average End-to-End Delay Raw Data (3 of 3)

Scenario	Seed									
	504	**311**	**1604**	**542**	**78**	**1819**	**357**	**737**	**698**	**892**
1	1552.08	0.61316	0.36082	0.96632	0.29367	0.16443	0.53748	2.60047	2.10086	0.71894
2	0.21549	0.09018	1.08622	0.11114	0.12106	0.22689	0.21068	0.05815	0.41389	0.09563
3	0.09921	0.15454	0.02173	0.09227	0.05261	0.17927	0.08144	0.07235	0.09578	0.04977
4	0.16543	0.03683	0.06316	0.05732	0.13365	0.06140	0.11562	0.06645	0.08983	0.05970
5	0.03285	0.05707	0.05289	0.10028	0.06382	0.01775	0.00875	0.05563	0.06787	0.03579
6	0.04404	0.01361	0.01236	0.01063	0.00576	0.01302	0.00707	0.01417	0.00790	0.03697
7	0.00282	0.00254	0.00223	0.00236	0.00339	0.00340	0.00267	0.00285	0.00285	0.00289
8	0.00148	0.00124	0.00154	0.00213	0.00190	0.00157	0.00154	0.00173	0.00180	0.00179
9	0.00136	0.00137	0.00144	0.00132	0.00112	0.00113	0.00128	0.00145	0.00146	0.00111
10	1037.20	0.37477	0.49283	1.35225	0.03810	0.25766	0.26800	0.68636	0.67701	0.42687
11	0.11345	0.08834	0.14606	0.11423	0.13734	0.07692	0.23690	0.03042	0.11775	0.07968
12	0.02080	0.02396	0.01374	0.05489	0.02474	0.04206	0.01871	0.05496	0.06566	0.06940
13	0.04049	0.08503	0.13645	0.08836	0.04460	0.05580	0.08202	0.06116	0.04563	0.07854
14	0.01357	0.01404	0.02263	0.01414	0.05944	0.02421	0.01030	0.01365	0.01467	0.03700
15	0.01195	0.00492	0.01587	0.00965	0.00385	0.00833	0.04671	0.01114	0.00930	0.00706
16	0.00177	0.00161	0.00197	0.00249	0.00227	0.00197	0.00179	0.00320	0.00314	0.00233
17	0.00180	0.00108	0.00188	0.00145	0.00134	0.00179	0.00148	0.00157	0.00133	0.00141
18	0.00117	0.00081	0.00140	0.00140	0.00126	0.00093	0.00149	0.00114	0.00127	0.00102

C.2 Packet Loss

Packet loss is calculated as the ratio of total VoIP packets received in the MANET versus the total number of packets sent. Each stream sends VoIP packets at a rate of 50 packets per second for all codec's. Each repetition is run for two hours (or 7200 seconds). The total number of VoIP packets sent for a scenario can be determined by

$$PacketsSent = 7200secs * 50PPS * STREAMS \qquad (2)$$

where *STREAMS* is the total number of streams in the scenario. Table 20, Table 21 and Table 22 show the total number of VoIP packets received.

Therefore, the total number of VoIP packets sent for scenarios with one, two and three streams are 360000, 720000 and 1080000 packets respectively.

Table 23, Table 24 and Table 25 show packet loss using the values collected from Table 20, Table 21 and Table 22 respectively. The percentage of packet loss observed is calculated by

$$Packet\ Loss = (1 - PacketsRecieved / PacketsSent) * 100\% \qquad (3)$$

where *PacketsRecieved* is the total number of VoIP packets received by the destination node(s) and *PacketsSent* is the total number of VoIP packets sent by the source for the duration of the simulation.

Table 20: Total Packets Received Raw Data (1 of 3)

Scenario	Seed									
	149	953	811	1019	37	17	397	683	277	977
1	992594	996079	1002179	973038	963860	1010935	995686	1004394	873048	989996
2	1024589	1036294	1039862	1034878	1033944	1026867	1028608	1035099	1024965	1026982
3	1052566	1026269	1035821	1036143	1038801	1044384	1041530	1037622	1045711	1049230
4	671052	666183	658444	669321	658746	685048	660453	664397	675209	669886
5	688287	678697	681497	689669	687339	691188	684363	693724	696128	676945
6	699816	691790	696203	690472	696025	691481	696658	689707	695371	701765
7	327026	333333	336805	330508	329924	335618	332192	329884	337614	340467
8	345355	349433	344359	350989	344666	347117	341610	339500	342580	336913
9	343962	346965	345795	347547	350579	347822	351365	348479	346580	345715
10	1027175	1015803	1020194	1020603	1012464	1026853	1019558	1025309	991411	1013994
11	1049494	1040324	1030552	1046552	1046799	1034642	1040919	1050452	1044828	1039504
12	1050261	1054682	1036469	1052089	1054644	1036970	1041256	1048358	1055655	1053399
13	682714	664939	684103	656473	692853	679901	676769	681138	671663	686978
14	690910	680295	690812	697552	695413	695223	698528	684411	694229	681290
15	701198	698022	703379	704576	698592	705880	707537	705585	705352	698923
16	336802	348728	336438	333199	342704	341760	333265	343276	341715	342329
17	349302	348719	347434	342356	343945	351144	343060	337301	351099	348163
18	349120	350202	352041	351700	352355	351862	341312	352370	349351	345773

Table 21: Total Packets Received Raw Data (2 of 3)

Scenario	Seed									
	2833	929	107	421	463	1511	809	449	1109	601
1	998359	984192	1013846	1013358	1013313	1006456	1021942	1002659	1030394	992366
2	1030396	1025479	1006319	1033310	1031873	1029436	1025408	1031767	1025381	1027367
3	1046906	1037437	1045633	1040408	1038159	1046142	1032396	1041299	1044666	1049884
4	679581	666225	675589	671311	662759	661276	666371	656930	655162	657004
5	687160	691607	686350	676564	683064	676816	682590	689452	680369	683371
6	697211	691572	698337	704793	696685	694752	697596	697006	690886	695837
7	337317	332356	340897	339398	324690	334726	338463	335186	332789	330626
8	340538	340805	343662	342023	340793	347674	335711	344463	346148	339477
9	349559	351184	346836	349358	347067	349358	346651	346195	348823	345918
10	1037970	1011553	1008702	994883	1025384	1000595	1018833	1001933	987159	1022984
11	1022112	1032361	1038919	1046888	1013260	1023593	1032747	1040655	1023930	1034216
12	1040217	1061110	1046206	1034341	1050362	1056503	1043603	1053477	1054906	1038162
13	648887	673273	681061	685403	688053	663871	675848	688045	674533	681598
14	697248	689192	686500	688450	693924	688736	689184	682582	693382	701708
15	696140	701316	703846	700973	697468	703302	698094	702778	700538	692797
16	339822	325752	340340	334528	331851	348372	342571	345142	341123	330220
17	347554	347489	341593	346913	346015	348080	347976	347507	345423	334817
18	354666	353733	350877	353050	345922	349780	351483	351733	347456	352687

Table 22: Total Packets Received Raw Data (3 of 3)

Scenario	Seed									
	504	**311**	**1604**	**542**	**78**	**1819**	**357**	**737**	**698**	**892**
1	298091	997220	1003327	1001714	1008515	1013830	998277	985363	982517	1002297
2	1018431	1038219	1022019	1039011	1025205	1008680	1033715	1031480	1015773	1030818
3	1043290	1031512	1051330	1035764	1042535	1033064	1040586	1049437	1042587	1040376
4	663506	677089	670719	671460	667731	675925	670568	676290	665808	674352
5	681025	679171	674275	680505	676048	683726	686120	683264	679670	696854
6	696663	694067	692701	694207	698331	697795	696442	692871	698344	697770
7	333885	337727	339955	341023	331975	335492	341369	331674	336063	334609
8	339956	348995	346763	341145	337741	344957	346375	345469	343439	343000
9	348079	345229	348289	347791	350408	349967	348369	347837	338355	351211
10	382897	1014847	995267	1013023	1034709	1012501	1014413	1007808	989854	1035841
11	1028428	1035565	1034354	1034641	1037905	1035240	1031538	1042155	1041427	1033671
12	1041595	1052492	1055737	1051202	1050537	1046577	1055436	1050645	1053307	1046364
13	676189	680356	677652	675658	677996	671503	673409	683930	670386	670076
14	693222	692426	688618	691745	692010	690080	685215	690342	694788	692350
15	703624	705992	694028	703155	709060	696451	691473	703466	689813	702141
16	346325	345388	339558	335994	340938	343076	339636	328619	335702	339854
17	339217	350959	342372	347127	347604	343968	345190	346268	343747	348308
18	342988	349443	348809	347841	350235	349809	347992	350980	348703	352179

Table 23: Packet Loss Raw Data (1 of 3)

Scenario	Seed									
	149	**953**	**811**	**1019**	**37**	**17**	**397**	**683**	**277**	**977**
1	8.09%	7.77%	7.21%	9.90%	10.75%	6.39%	7.81%	7.00%	19.16%	8.33%
2	5.13%	4.05%	3.72%	4.18%	4.26%	4.92%	4.76%	4.16%	5.10%	4.91%
3	2.54%	4.98%	4.09%	4.06%	3.81%	3.30%	3.56%	3.92%	3.17%	2.85%
4	6.80%	7.47%	8.55%	7.04%	8.51%	4.85%	8.27%	7.72%	6.22%	6.96%
5	4.40%	5.74%	5.35%	4.21%	4.54%	4.00%	4.95%	3.65%	3.32%	5.98%
6	2.80%	3.92%	3.31%	4.10%	3.33%	3.96%	3.24%	4.21%	3.42%	2.53%
7	9.16%	7.41%	6.44%	8.19%	8.35%	6.77%	7.72%	8.37%	6.22%	5.43%
8	4.07%	2.94%	4.34%	2.50%	4.26%	3.58%	5.11%	5.69%	4.84%	6.41%
9	4.46%	3.62%	3.95%	3.46%	2.62%	3.38%	2.40%	3.20%	3.73%	3.97%
10	4.89%	5.94%	5.54%	5.50%	6.25%	4.92%	5.60%	5.06%	8.20%	6.11%
11	2.82%	3.67%	4.58%	3.10%	3.07%	4.20%	3.62%	2.74%	3.26%	3.75%
12	2.75%	2.34%	4.03%	2.58%	2.35%	3.98%	3.59%	2.93%	2.25%	2.46%
13	5.18%	7.65%	4.99%	8.82%	3.77%	5.57%	6.00%	5.40%	6.71%	4.59%
14	4.04%	5.51%	4.05%	3.12%	3.41%	3.44%	2.98%	4.94%	3.58%	5.38%
15	2.61%	3.05%	2.31%	2.14%	2.97%	1.96%	1.73%	2.00%	2.03%	2.93%
16	6.44%	3.13%	6.55%	7.44%	4.80%	5.07%	7.43%	4.65%	5.08%	4.91%
17	2.97%	3.13%	3.49%	4.90%	4.46%	2.46%	4.71%	6.31%	2.47%	3.29%
18	3.02%	2.72%	2.21%	2.31%	2.12%	2.26%	5.19%	2.12%	2.96%	3.95%

Table 24: Packet Loss Raw Data (2 of 3)

Scenario	Seed									
	2833	929	107	421	463	1511	809	449	1109	601
1	7.56%	8.87%	6.13%	6.17%	6.17%	6.81%	5.38%	7.16%	4.59%	8.11%
2	4.59%	5.05%	6.82%	4.32%	4.46%	4.68%	5.05%	4.47%	5.06%	4.87%
3	3.06%	3.94%	3.18%	3.67%	3.87%	3.14%	4.41%	3.58%	3.27%	2.79%
4	5.61%	7.47%	6.17%	6.76%	7.95%	8.16%	7.45%	8.76%	9.01%	8.75%
5	4.56%	3.94%	4.67%	6.03%	5.13%	6.00%	5.20%	4.24%	5.50%	5.09%
6	3.17%	3.95%	3.01%	2.11%	3.24%	3.51%	3.11%	3.19%	4.04%	3.36%
7	6.30%	7.68%	5.31%	5.72%	9.81%	7.02%	5.98%	6.89%	7.56%	8.16%
8	5.41%	5.33%	4.54%	4.99%	5.34%	3.42%	6.75%	4.32%	3.85%	5.70%
9	2.90%	2.45%	3.66%	2.96%	3.59%	2.96%	3.71%	3.83%	3.10%	3.91%
10	3.89%	6.34%	6.60%	7.88%	5.06%	7.35%	5.66%	7.23%	8.60%	5.28%
11	5.36%	4.41%	3.80%	3.07%	6.18%	5.22%	4.38%	3.64%	5.19%	4.24%
12	3.68%	1.75%	3.13%	4.23%	2.74%	2.18%	3.37%	2.46%	2.32%	3.87%
13	9.88%	6.49%	5.41%	4.81%	4.44%	7.80%	6.13%	4.44%	6.31%	5.33%
14	3.16%	4.28%	4.65%	4.38%	3.62%	4.34%	4.28%	5.20%	3.70%	2.54%
15	3.31%	2.60%	2.24%	2.64%	3.13%	2.32%	3.04%	2.39%	2.70%	3.78%
16	5.61%	9.51%	5.46%	7.08%	7.82%	3.23%	4.84%	4.13%	5.24%	8.27%
17	3.46%	3.48%	5.11%	3.64%	3.88%	3.31%	3.34%	3.47%	4.05%	7.00%
18	1.48%	1.74%	2.53%	1.93%	3.91%	2.84%	2.37%	2.30%	3.48%	2.03%

Table 25: Packet Loss Raw Data (3 of 3)

Scenario	Seed									
	504	311	1604	542	78	1819	357	737	698	892
1	72.40%	7.66%	7.10%	7.25%	6.62%	6.13%	7.57%	8.76%	9.03%	7.19%
2	5.70%	3.87%	5.37%	3.80%	5.07%	6.60%	4.29%	4.49%	5.95%	4.55%
3	3.40%	4.49%	2.65%	4.10%	3.47%	4.35%	3.65%	2.83%	3.46%	3.67%
4	7.85%	5.96%	6.84%	6.74%	7.26%	6.12%	6.87%	6.07%	7.53%	6.34%
5	5.41%	5.67%	6.35%	5.49%	6.10%	5.04%	4.71%	5.10%	5.60%	3.21%
6	3.24%	3.60%	3.79%	3.58%	3.01%	3.08%	3.27%	3.77%	3.01%	3.09%
7	7.25%	6.19%	5.57%	5.27%	7.78%	6.81%	5.18%	7.87%	6.65%	7.05%
8	5.57%	3.06%	3.68%	5.24%	6.18%	4.18%	3.78%	4.04%	4.60%	4.72%
9	3.31%	4.10%	3.25%	3.39%	2.66%	2.79%	3.23%	3.38%	6.01%	2.44%
10	64.55%	6.03%	7.85%	6.20%	4.19%	6.25%	6.07%	6.68%	8.35%	4.09%
11	4.78%	4.11%	4.23%	4.20%	3.90%	4.14%	4.49%	3.50%	3.57%	4.29%
12	3.56%	2.55%	2.25%	2.67%	2.73%	3.09%	2.27%	2.72%	2.47%	3.11%
13	6.08%	5.51%	5.88%	6.16%	5.83%	6.74%	6.47%	5.01%	6.89%	6.93%
14	3.72%	3.83%	4.36%	3.92%	3.89%	4.16%	4.83%	4.12%	3.50%	3.84%
15	2.27%	1.95%	3.61%	2.34%	1.52%	3.27%	3.96%	2.30%	4.19%	2.48%
16	3.80%	4.06%	5.68%	6.67%	5.30%	4.70%	5.66%	8.72%	6.75%	5.60%
17	5.77%	2.51%	4.90%	3.58%	3.44%	4.45%	4.11%	3.81%	4.51%	3.25%
18	4.73%	2.93%	3.11%	3.38%	2.71%	2.83%	3.34%	2.51%	3.14%	2.17%

Appendix D. Analysis

This appendix presents an extensive analysis of the raw data collected. Section D.1 shows the Data Assumption Analysis to include Normal Q-Q plots, Residual Distributions, Versus Fits and Versus order plots. Section D.2 shows the 90% CI for the mean of the data collected. Section D.3 shows the *p*-values using a *t*-test between all combinations of scenarios.

D.1 Data Assumption Analysis

D.1.1 Normal Q-Q Plot

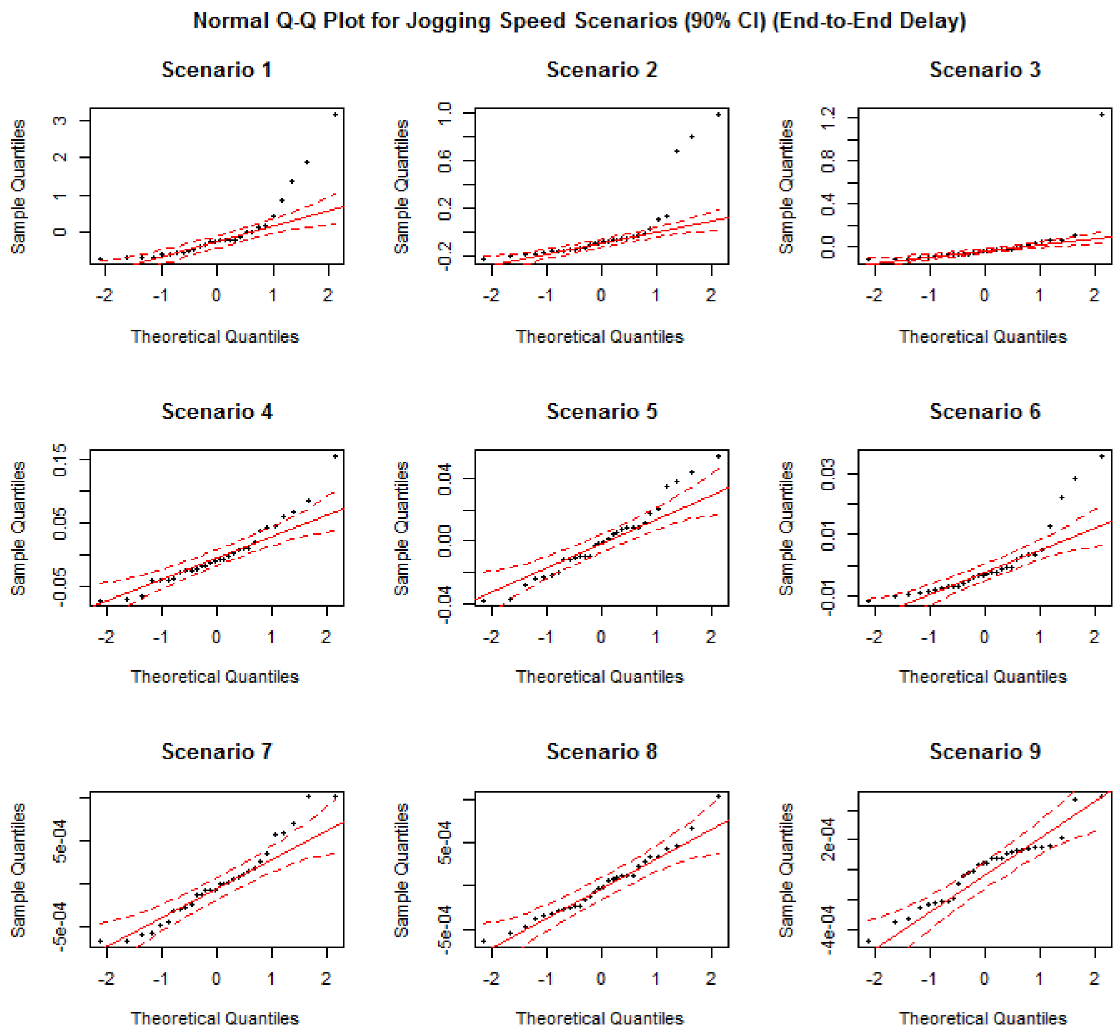


Figure 39: Normal Q-Q Plots for Jogging Speed (End-to-End Delay)

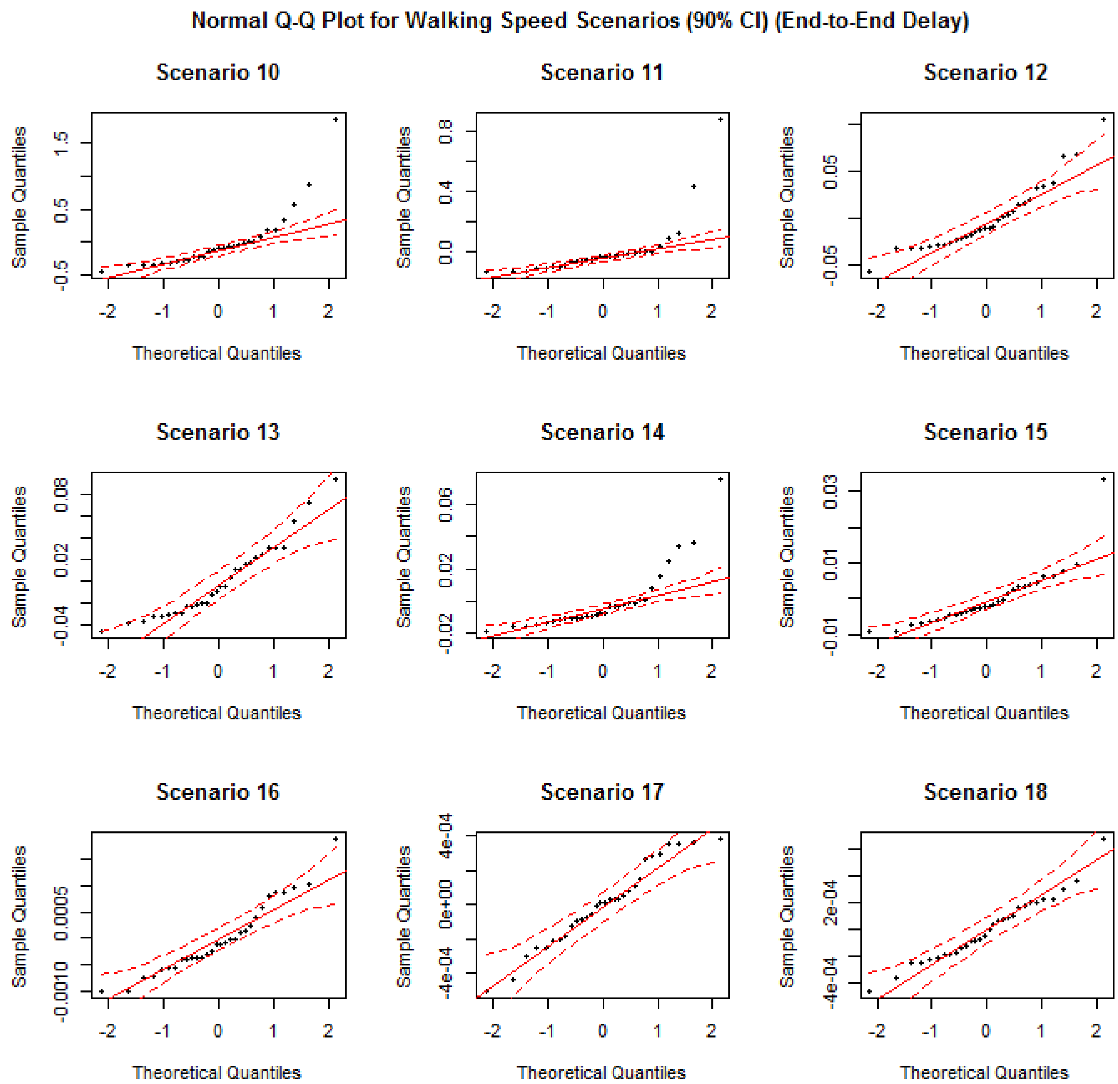


Figure 40: Normal Q-Q Plots for Walking Speed (End-to-End Delay)

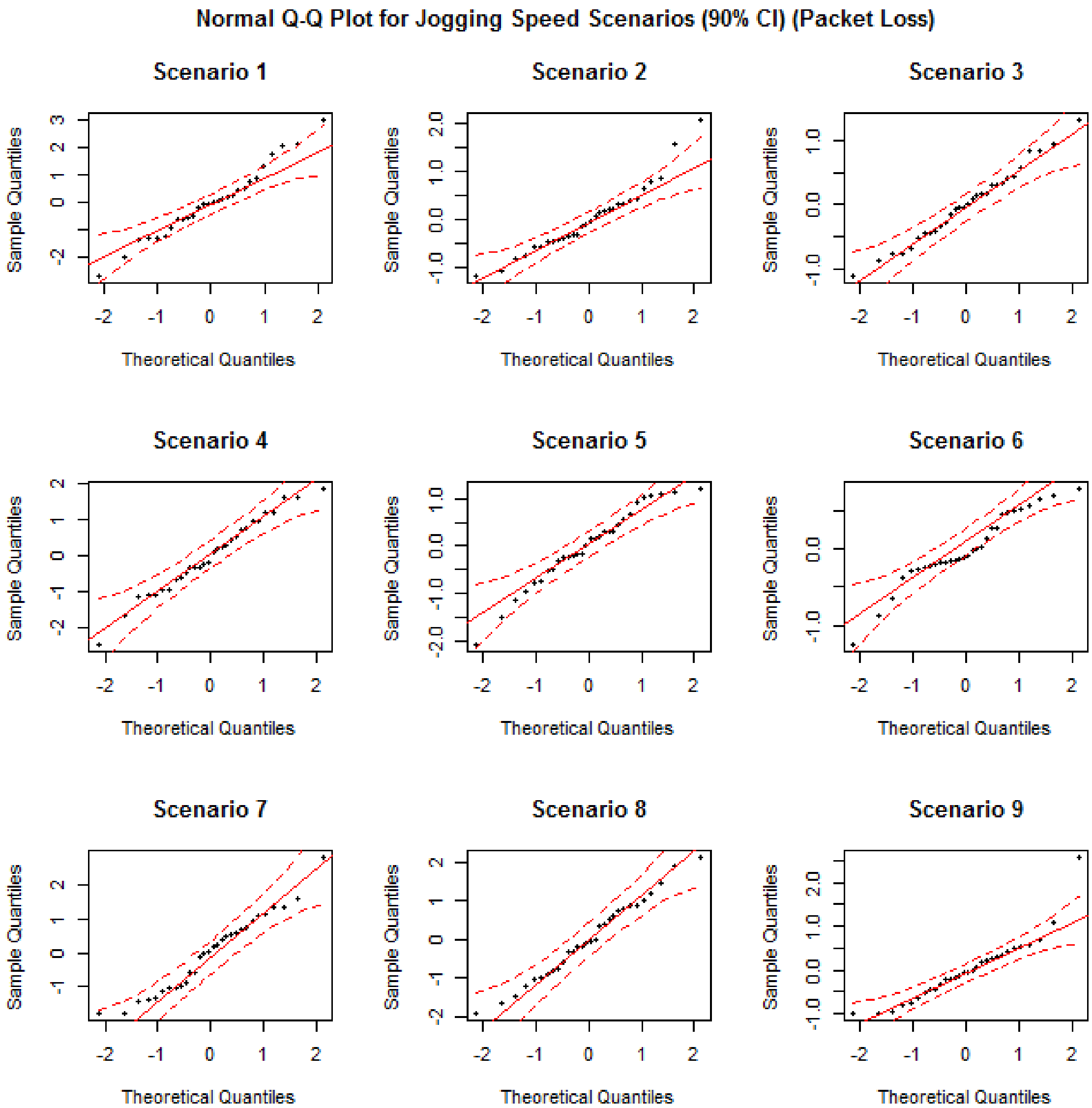


Figure 41: Normal Q-Q Plots for Jogging Speed (Packet Loss)

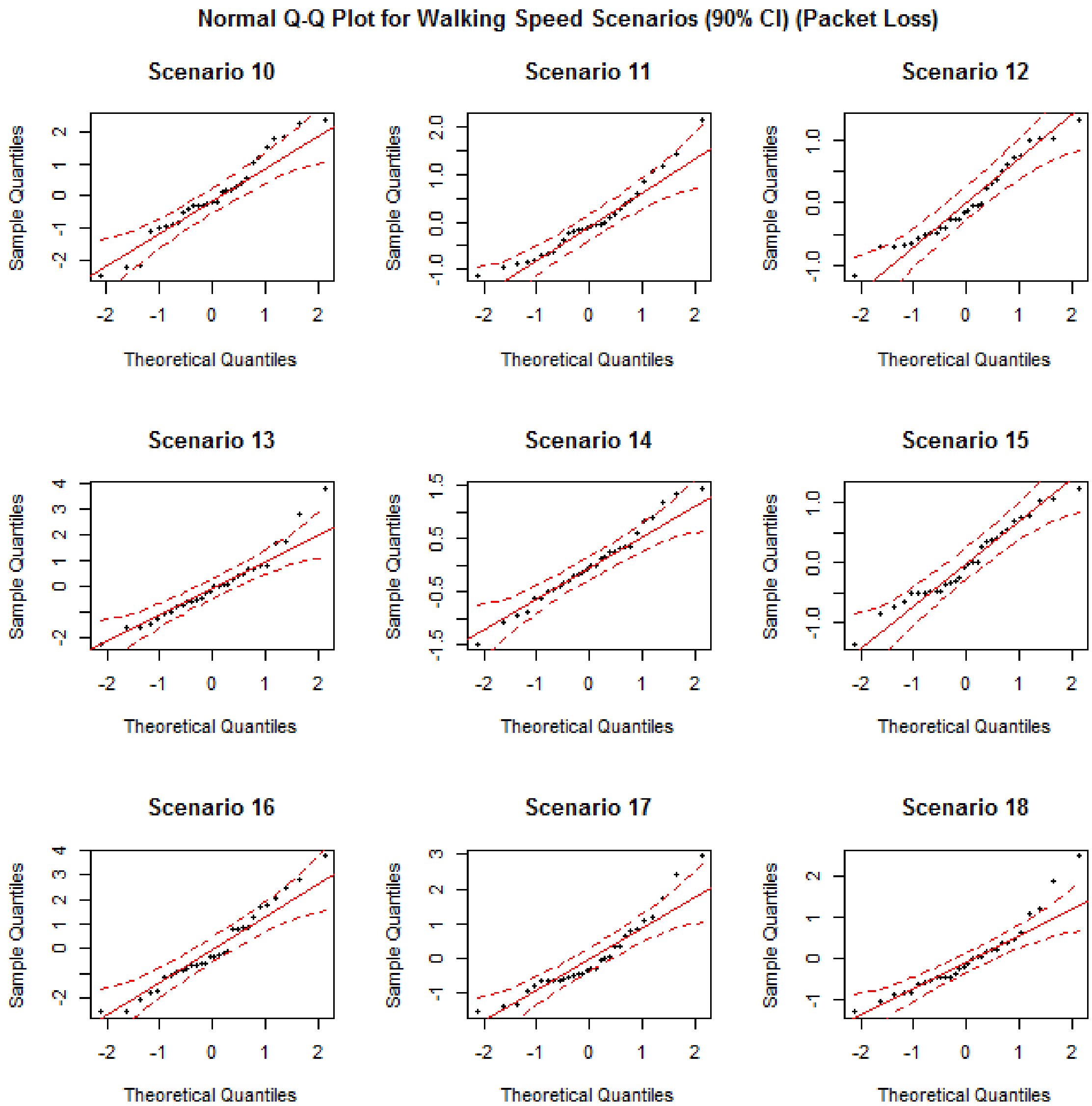


Figure 42: Normal Q-Q Plots for Walking Speed (Packet Loss)

D.1.2 Residual Distributions

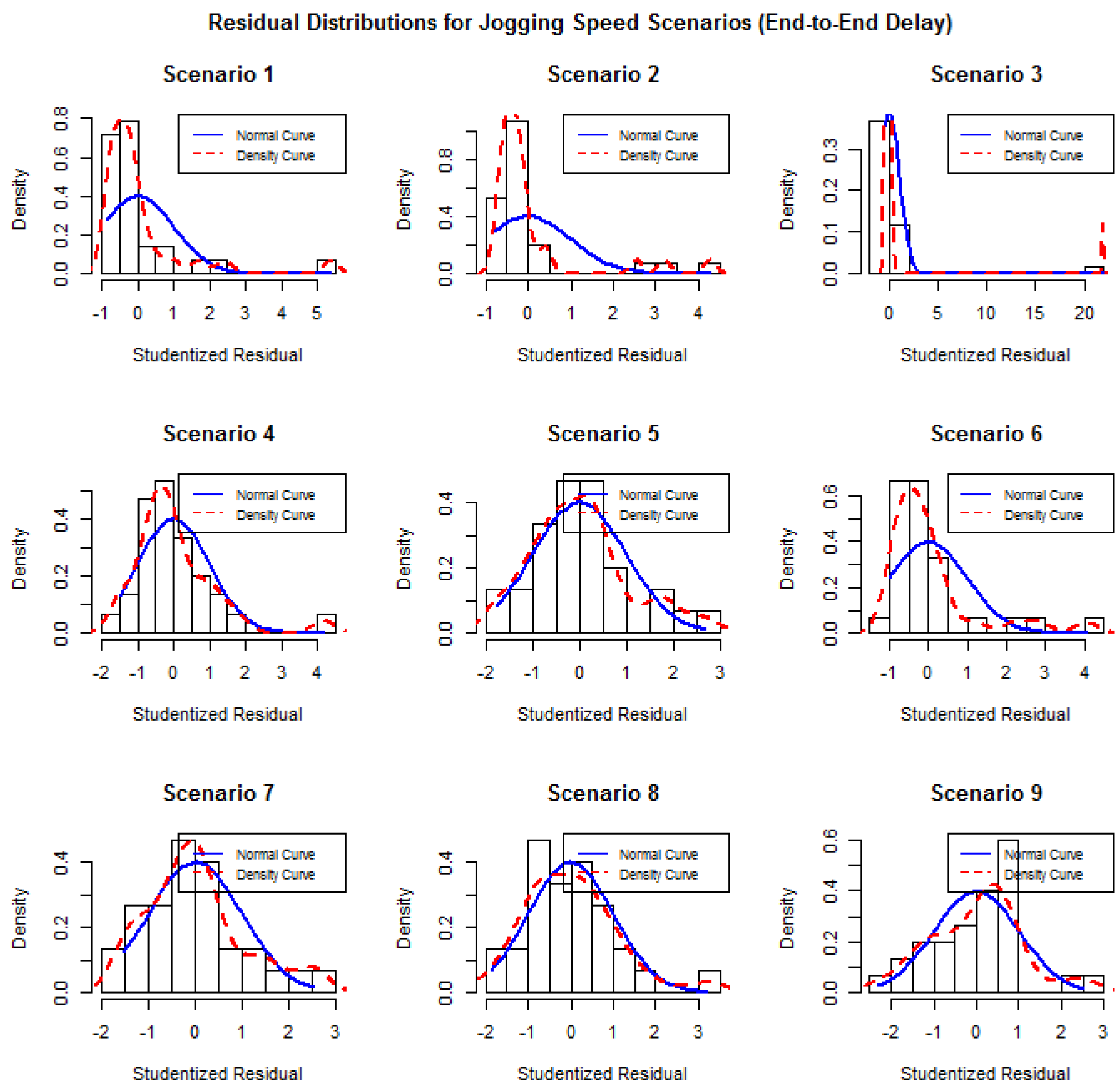


Figure 43: Residual Distributions for Jogging Speed (End-to-End Delay)

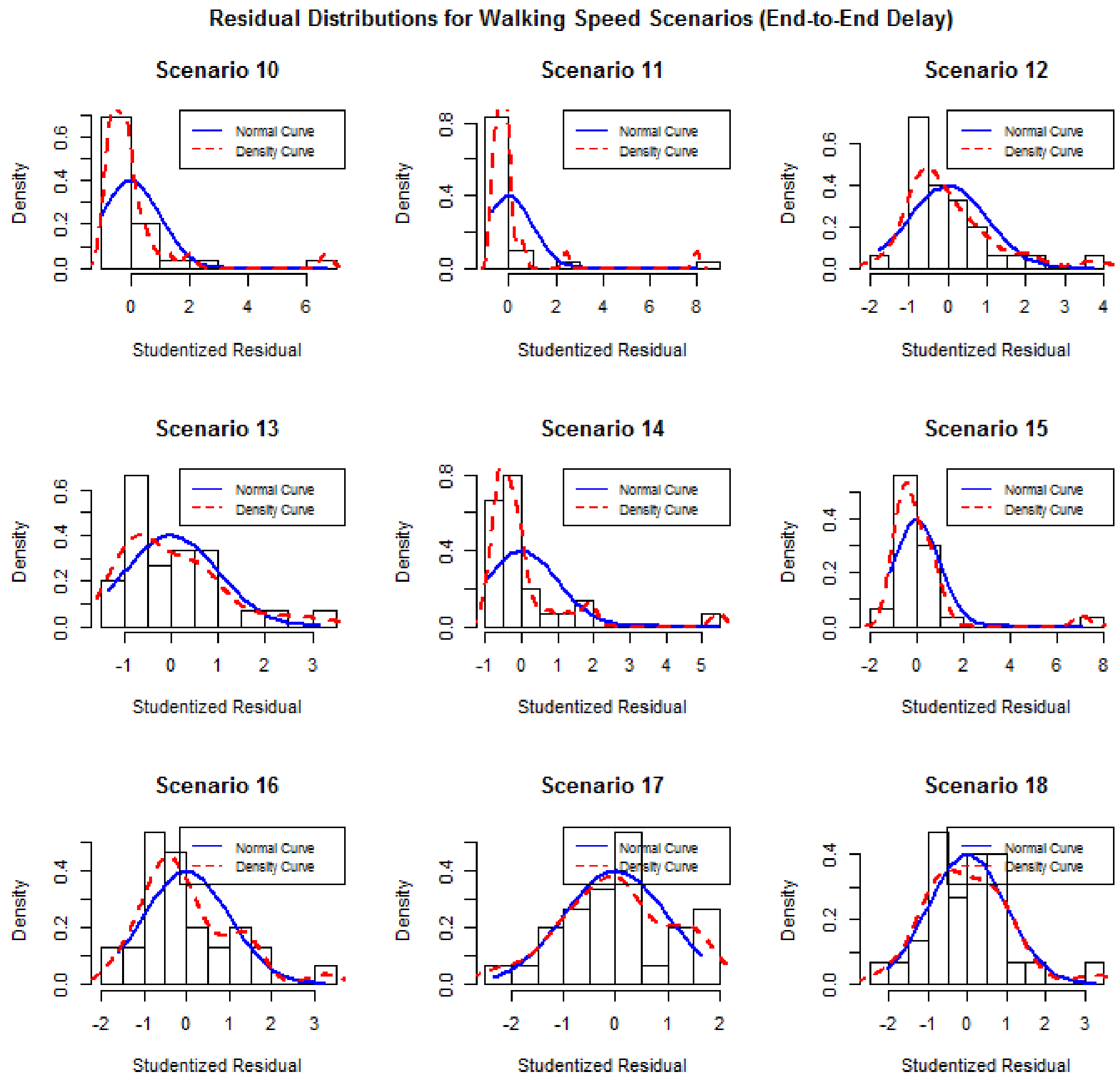


Figure 44: Residual Distributions for Walking Speed (End-to-End Delay)

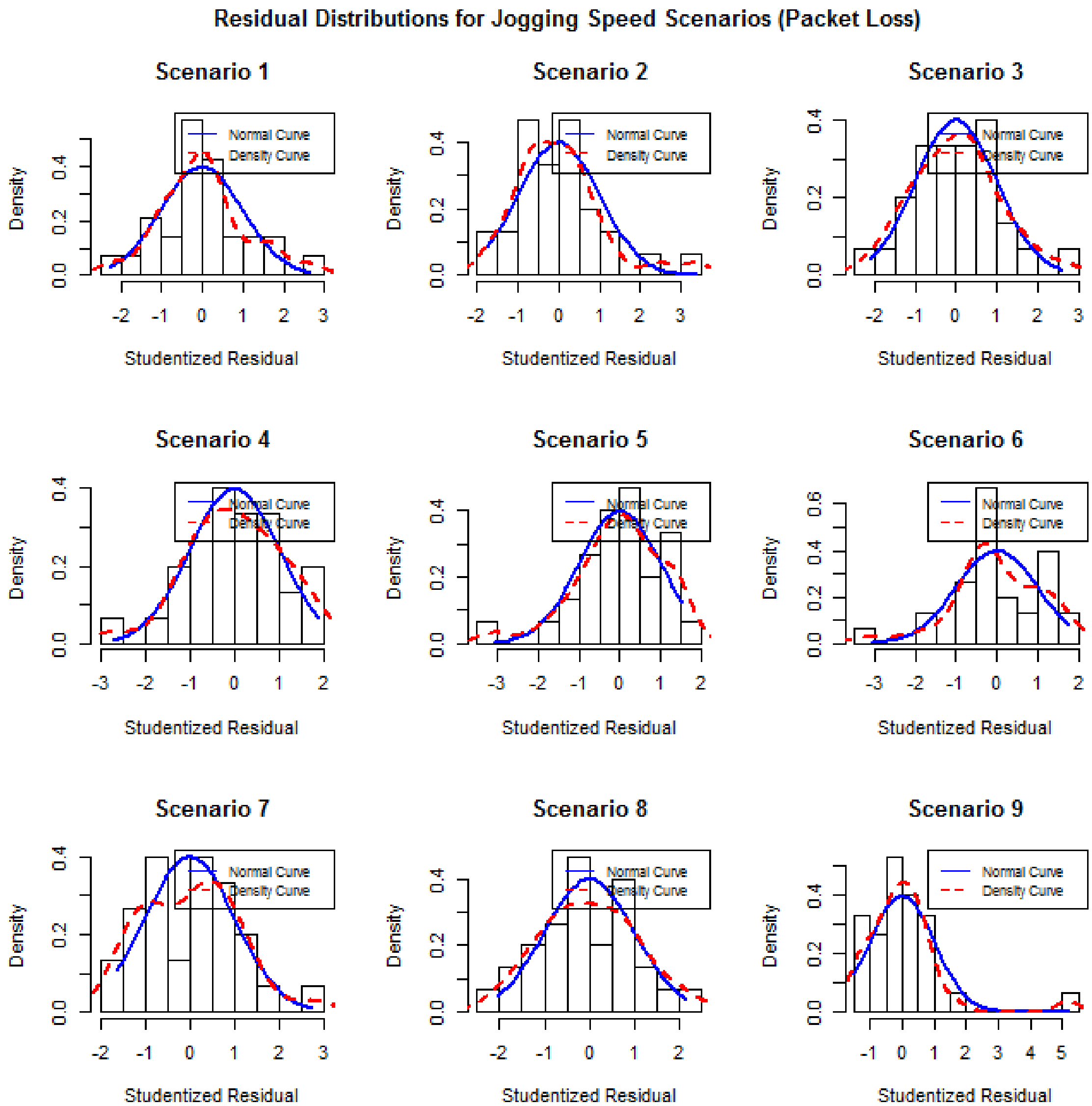


Figure 45: Residual Distributions for Jogging Speed (Packet Loss)

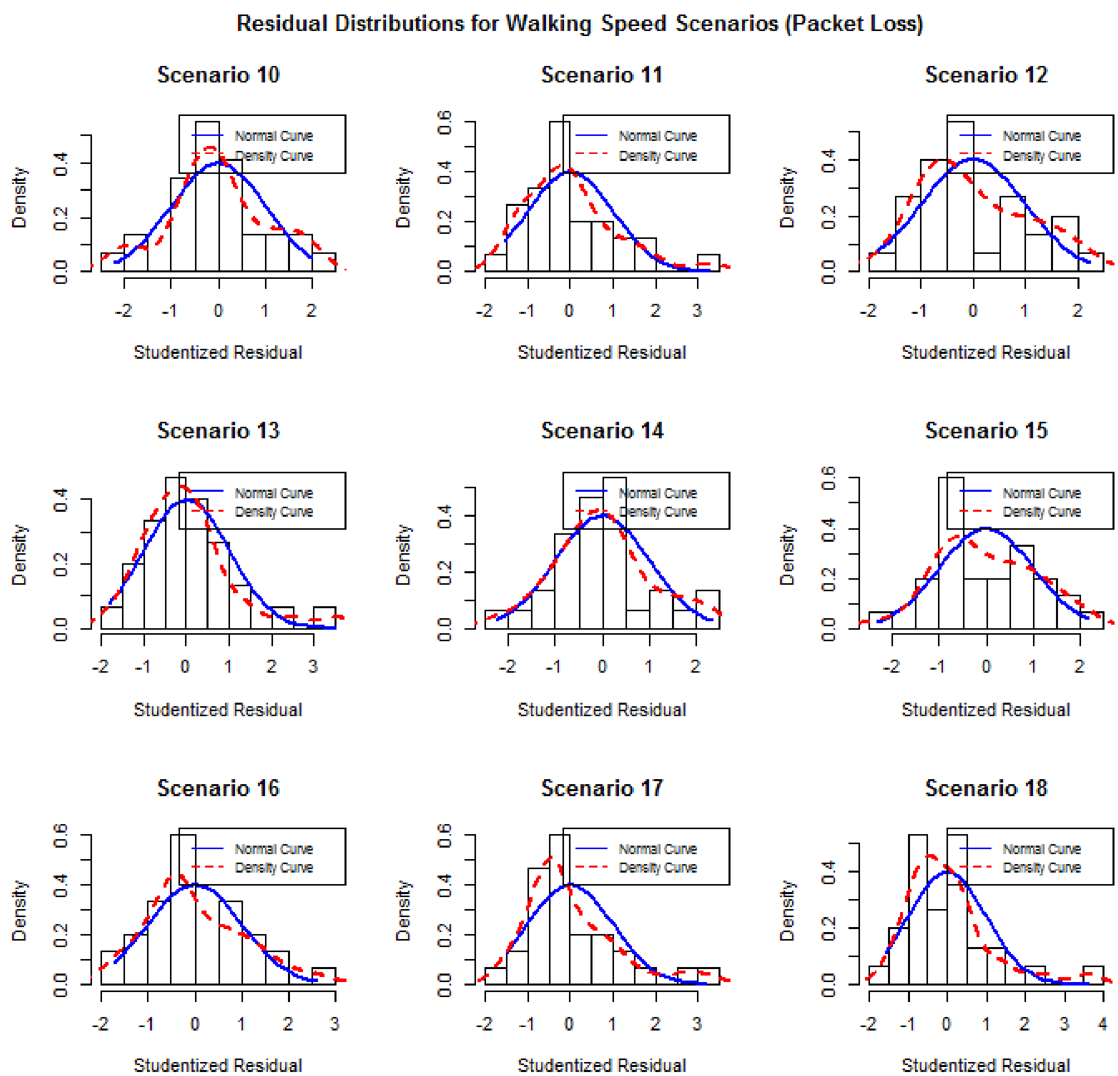


Figure 46: Residual Distributions for Walking Speed (Packet Loss)

D.1.3 Versus Fits

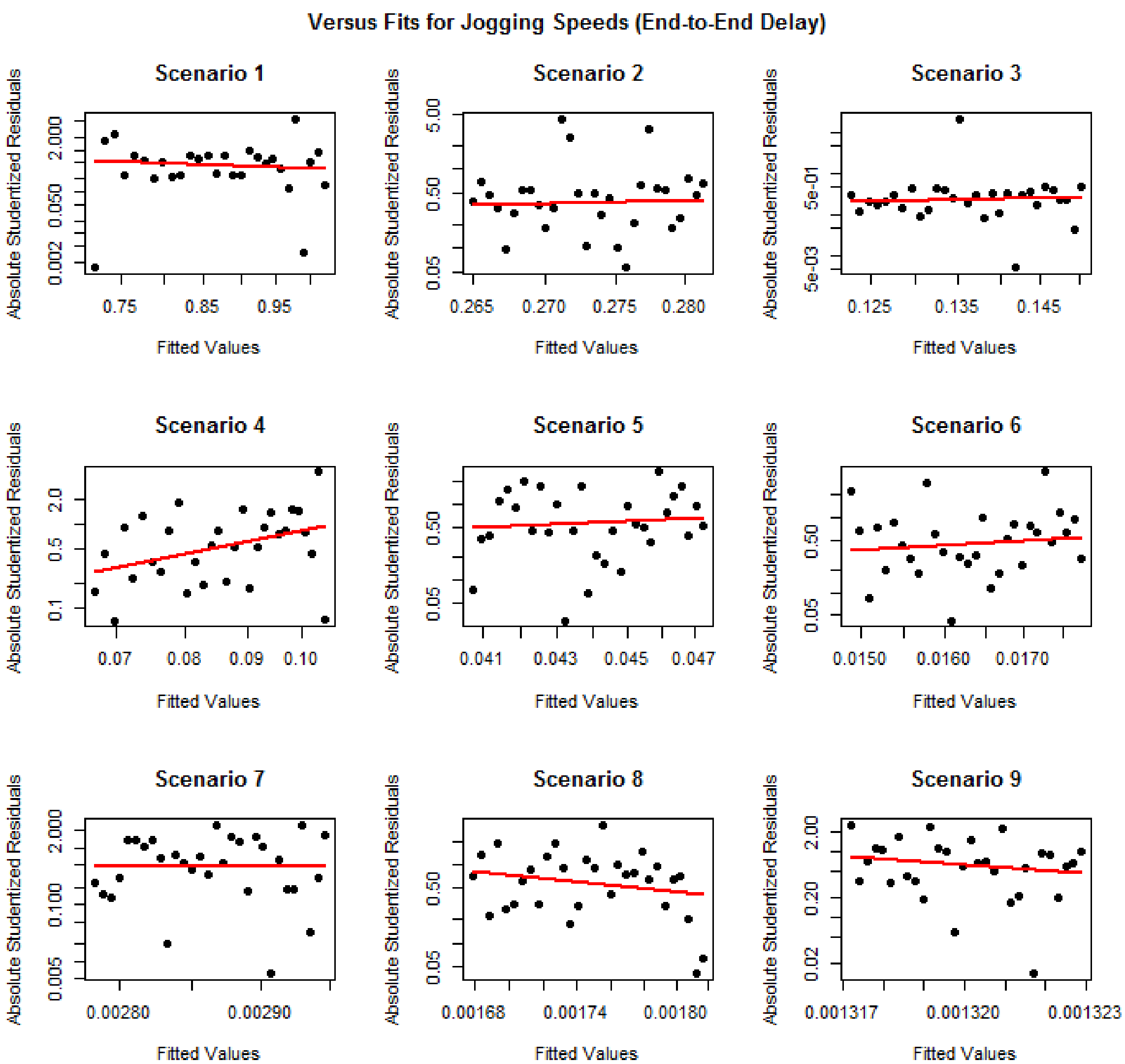


Figure 47: Versus Fits for Jogging Speed (End-to-End Delay)

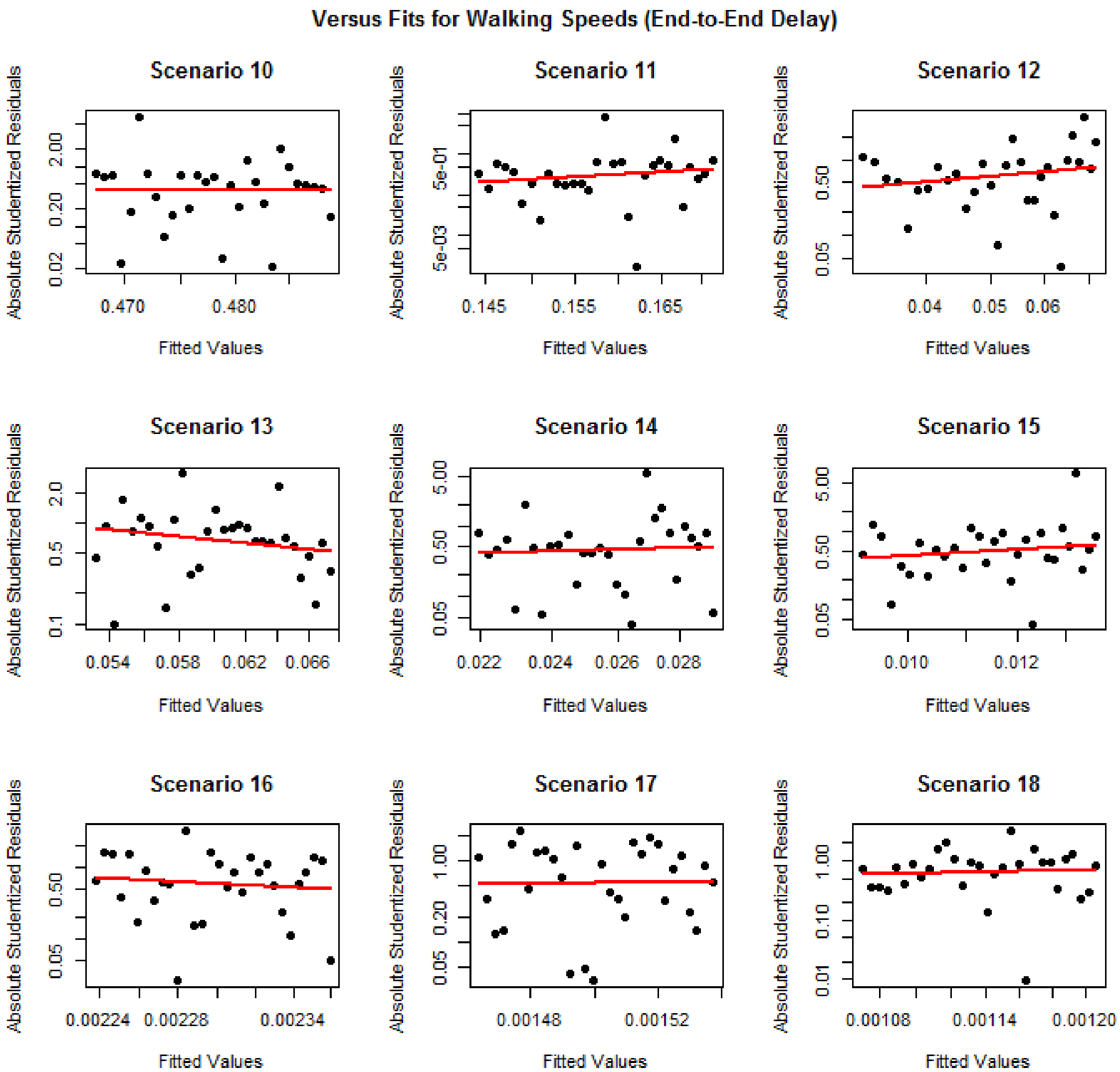


Figure 48: Versus Fits for Walking Speed (End-to-End Delay)

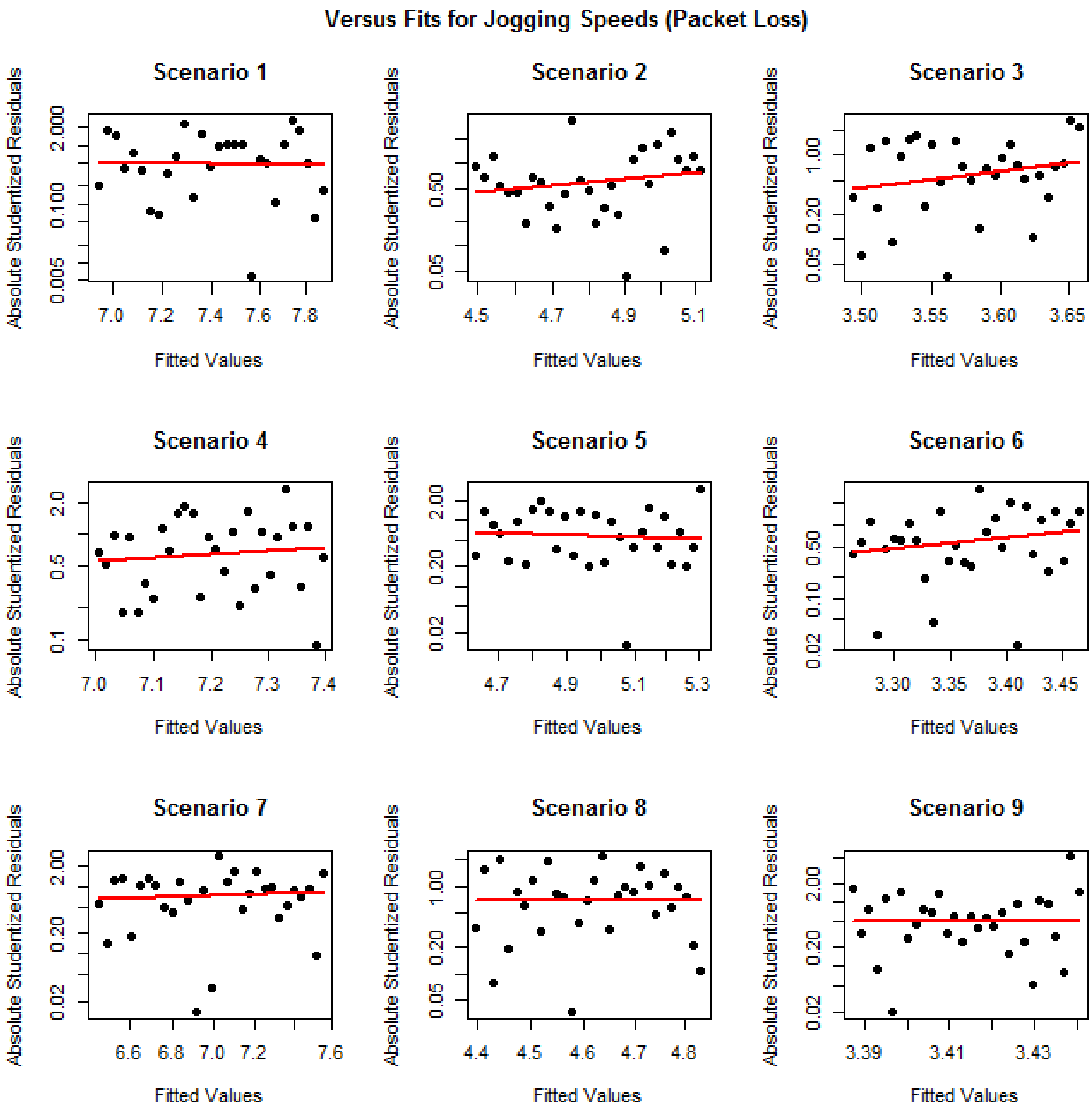


Figure 49: Versus Fits for Jogging Speed (Packet Loss)

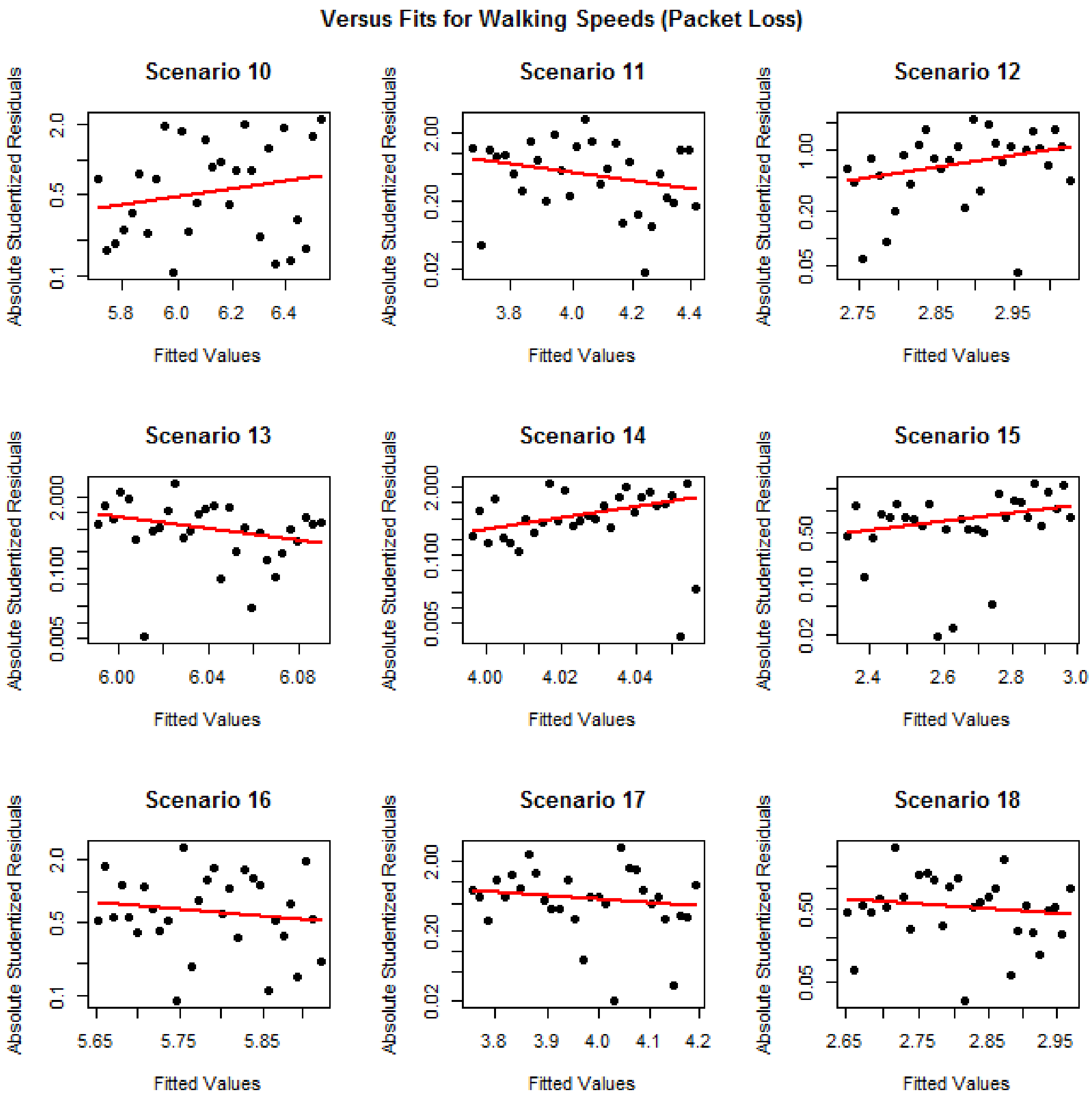


Figure 50: Versus Fits for Walking Speed (Packet Loss)

D.1.4 Versus Order

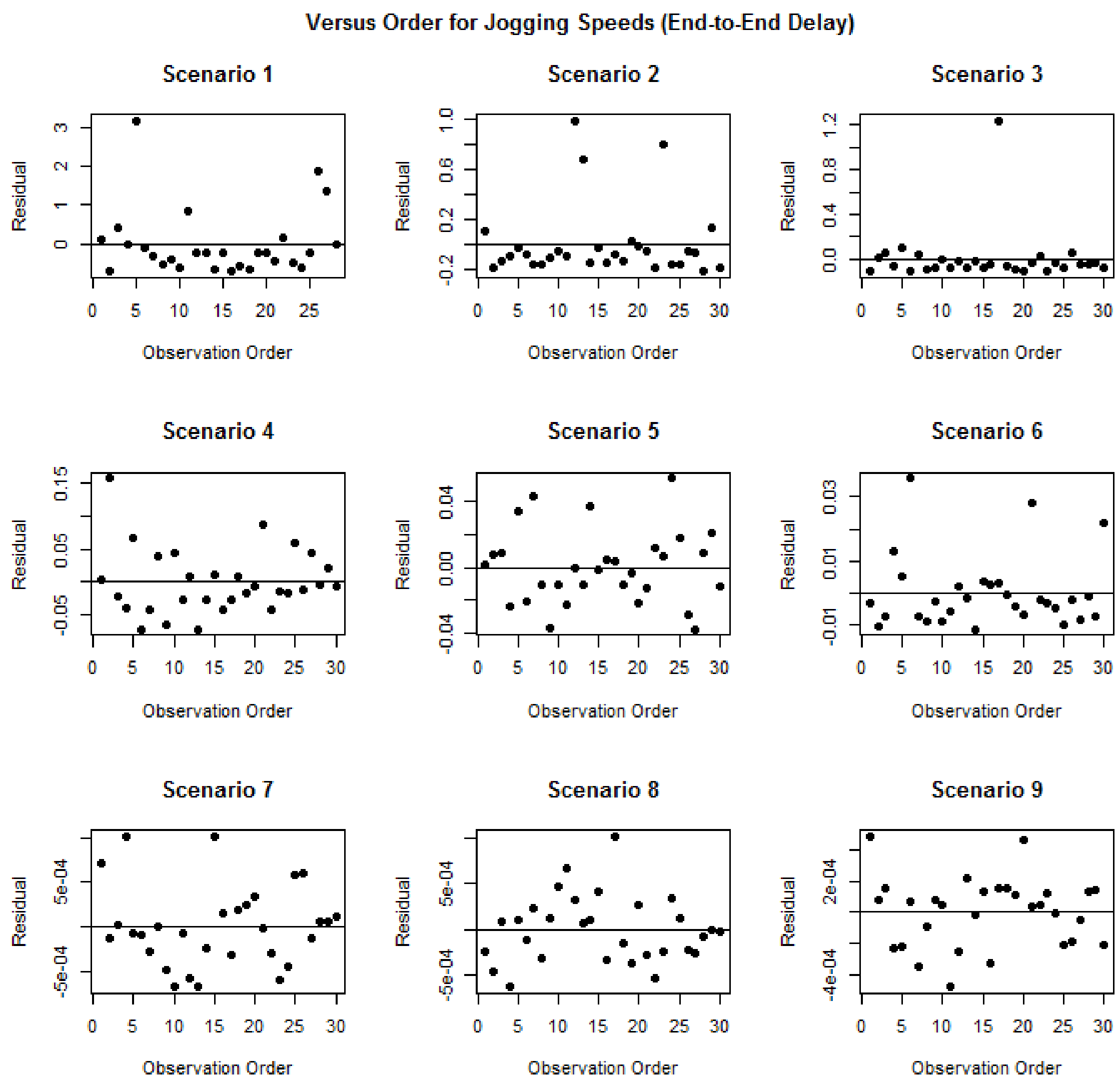


Figure 51: Versus Order for Jogging Speed (End-to-End Delay)

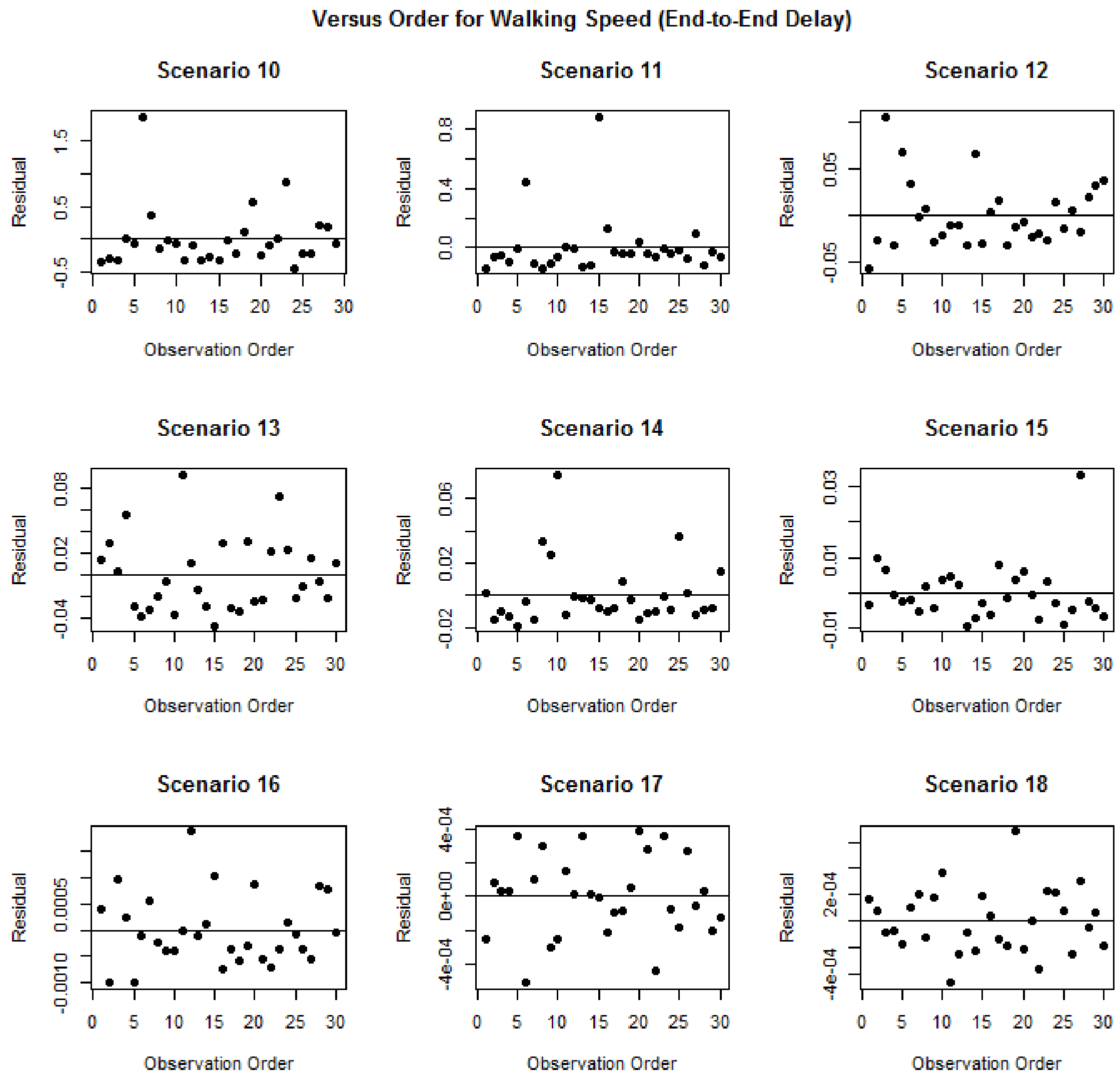


Figure 52: Versus Order for Walking Speed (End-to-End Delay)

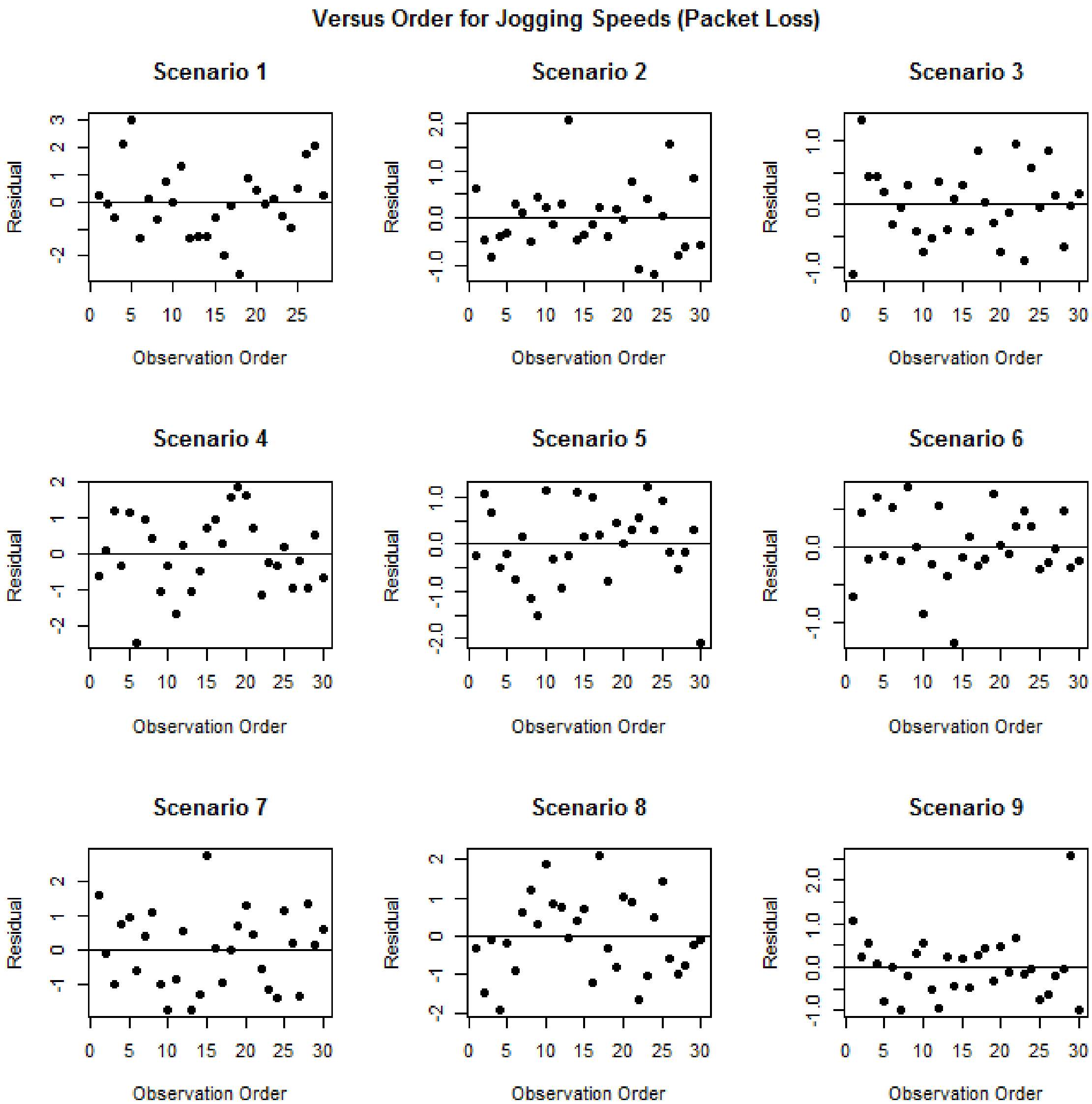


Figure 53: Versus Order for Jogging Speed (Packet Loss)

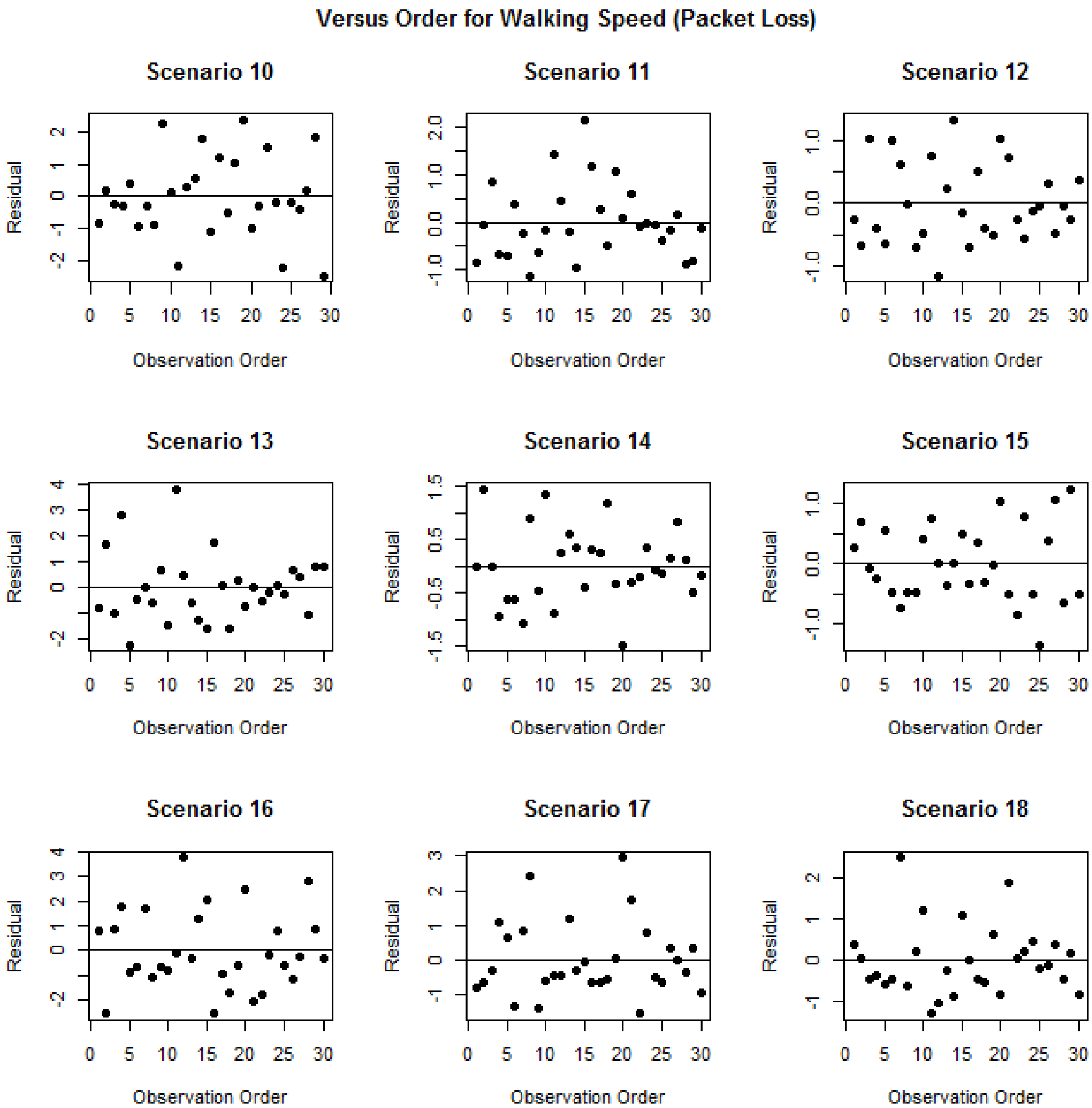


Figure 54: Versus Order for Walking Speed (Packet Loss)

D.2 90% Confidence Intervals

Table 26: Confidence Intervals for All Scenarios (End-to-End Delay)

Scenario #	Scenario Name	n	Mean	St Dev	Min	Max	Lower CI	Upper CI
1	2.5_3_200	28	0.872122953	0.87212	0.16443	4.1321	0.601003	1.143243
2	2.5_3_120	30	0.273142646	0.29582	0.05815	1.2631	0.184299	0.361987
3	2.5_3_60	30	0.136603947	0.24049	0.01934	1.37283	0.064377	0.208831
4	2.5_2_200	30	0.085940351	0.05077	0.01727	0.26025	0.070693	0.101187
5	2.5_2_120	30	0.044004368	0.02285	0.00508	0.10028	0.037143	0.050866
6	2.5_2_60	30	0.016328313	0.01125	0.00514	0.053	0.012949	0.019707
7	2.5_1_200	30	0.002864582	0.00045	0.0022	0.00394	0.002728	0.003001
8	2.5_1_120	30	0.001747996	0.00037	0.00105	0.00279	0.001636	0.00186
9	2.5_1_60	30	0.001320053	0.00022	0.00084	0.0018	0.001253	0.001387
10	1.5_3_200	29	0.478115435	0.4598	0.0381	2.33283	0.337661	0.61857
11	1.5_3_120	30	0.157906783	0.19743	0.02789	1.03493	0.098611	0.217203
12	1.5_3_60	30	0.051966998	0.03747	0.01374	0.17435	0.040712	0.063222
13	1.5_2_200	30	0.060373562	0.03437	0.01389	0.15139	0.05005	0.070697
14	1.5_2_120	30	0.025569192	0.01986	0.00943	0.10208	0.019604	0.031535
15	1.5_2_60	30	0.011467303	0.00822	0.00182	0.04671	0.008998	0.013937
16	1.5_1_200	30	0.002298795	0.00067	0.00124	0.00415	0.002098	0.0025
17	1.5_1_120	30	0.00150097	0.00024	0.00097	0.00189	0.001429	0.001573
18	1.5_1_60	30	0.001138419	0.00025	0.00065	0.00183	0.001065	0.001212

Table 27: Confidence Intervals for All Scenarios (Packet Loss)

Scenario #	Scenario Name	n	Mean	St Dev	Min	Max	Lower CI	Upper CI
1	2.5_3_200	28	7.411686508	1.3205	4.59315	10.7537	7.001176	7.822197
2	2.5_3_120	30	4.808101852	0.7411	3.71648	6.82231	4.585524	5.030679
3	2.5_3_60	30	3.575654321	0.58411	2.54019	4.97509	3.400227	3.751082
4	2.5_2_200	30	7.201875	1.0322	4.85444	9.00528	6.891871	7.511879
5	2.5_2_120	30	4.972972222	0.83805	3.21472	6.35069	4.721276	5.224669
6	2.5_2_60	30	3.365027778	0.47316	2.11208	4.20736	3.222922	3.507134
7	2.5_1_200	30	7.003787037	1.18309	5.17528	9.80833	6.648465	7.359109
8	2.5_1_120	30	4.614296296	1.0416	2.50306	6.74694	4.301468	4.927125
9	2.5_1_60	30	3.413953704	0.72019	2.39861	6.0125	3.197654	3.630253
10	1.5_3_200	29	6.125086207	1.26552	3.89167	8.59639	5.738509	6.511663
11	1.5_3_120	30	4.050398148	0.79277	2.73593	6.17963	3.812302	4.288494
12	1.5_3_60	30	2.880981481	0.64378	1.74907	4.22769	2.687631	3.074332
13	1.5_2_200	30	6.040472222	1.31053	3.77042	9.87681	5.646874	6.434071
14	1.5_2_120	30	4.026087963	0.70256	2.54056	5.51458	3.815086	4.23709
15	1.5_2_60	30	2.659726852	0.6656	1.51944	4.19264	2.459825	2.859628
16	1.5_1_200	30	5.786787037	1.58379	3.13111	9.51333	5.311121	6.262454
17	1.5_1_120	30	3.975462963	1.08205	2.46	6.99528	3.650487	4.300439
18	1.5_1_60	30	2.81062963	0.83855	1.48167	5.19111	2.558785	3.062474

D.3 *p*-Values (*t*-test)

Table 28: *p*-Values for End-to-End Delay (1 of 3)

Compare	Lower	Upper	P-value	Compare	Lower	Upper	P-value	Compare	Lower	Upper	*p* -Value
1-2	0.3054	0.89256	0.00155	**2-6**	0.16499	0.34864	5E-05	**3-11**	-0.11632	0.073713	0.7091
1-3	0.44628	1.02476	0.00015	**2-7**	0.17851	0.36205	2.5E-05	**3-12**	0.009248	0.160026	0.06632
1-4	0.50507	1.06729	5.7E-05	**2-8**	0.17963	0.36316	2.4E-05	**3-13**	0.000966	0.151495	0.09591
1-5	0.54731	1.10892	2.9E-05	**2-9**	0.18006	0.36359	2.3E-05	**3-14**	0.03621	0.185859	0.0174
1-6	0.57505	1.13654	1.8E-05	**2-10**	-0.3745	-0.0355	0.0481	**3-15**	0.050495	0.199778	0.00799
1-7	0.58853	1.14999	1.5E-05	**2-11**	0.00644	0.22403	0.08197	**3-16**	0.059701	0.208909	0.00475
1-8	0.58965	1.1511	1.4E-05	**2-12**	0.12877	0.31358	0.00032	**3-17**	0.060499	0.209707	0.00453
1-9	0.59008	1.15153	1.4E-05	**2-13**	0.12046	0.30507	0.00049	**3-18**	0.060862	0.210069	0.00444
1-10	0.08157	0.70645	0.03992	**2-14**	0.15563	0.33952	8.1E-05	**4-5**	0.024824	0.059048	0.00018
1-11	0.42774	1.00069	0.0002	**2-15**	0.16988	0.35347	3.9E-05	**4-6**	0.053529	0.085695	2.5E-08
1-12	0.53922	1.10109	3.3E-05	**2-16**	0.17908	0.36261	2.4E-05	**4-7**	0.067327	0.098825	7.4E-10
1-13	0.53085	1.09265	3.7E-05	**2-17**	0.17987	0.36341	2.3E-05	**4-8**	0.068443	0.099941	5.6E-10
1-14	0.56577	1.12734	2.1E-05	**2-18**	0.18024	0.36377	2.3E-05	**4-9**	0.068871	0.100369	5E-10
1-15	0.57992	1.14139	1.7E-05	**3-4**	-0.0254	0.12671	0.2674	**4-10**	-0.53816	-0.24619	8.7E-05
1-16	0.5891	1.15055	1.4E-05	**3-5**	0.0177	0.1675	0.04443	**4-11**	-0.13496	-0.00897	0.06182
1-17	0.5899	1.15135	1.4E-05	**3-6**	0.0456	0.19495	0.01047	**4-12**	0.014689	0.053257	0.00472
1-18	0.59026	1.15171	1.4E-05	**3-7**	0.05914	0.20834	0.0049	**4-13**	0.006814	0.044319	0.02657
2-3	0.02011	0.25296	0.05481	**3-8**	0.06025	0.20946	0.0046	**4-14**	0.043587	0.077155	4.8E-07
2-4	0.09426	0.28014	0.00181	**3-9**	0.06068	0.20989	0.00449	**4-15**	0.058545	0.090401	6.6E-09
2-5	0.13713	0.32114	0.00021	**3-10**	-0.503	-0.18	0.00095	**4-16**	0.067892	0.099391	6.4E-10

Table 29: *p*-Values for End-to-End Delay (2 of 3)

Compare	Lower	Upper	P-value	Compare	Lower	Upper	P-value	Compare	Lower	Upper	*p* -Value
4-17	0.06869	0.10019	5.2E-10	**6-12**	-0.0477	-0.0236	1.8E-05	**8-11**	-0.21741	-0.09491	0.00016
4-18	0.06905	0.10055	4.8E-10	**6-13**	-0.0552	-0.0329	1E-07	**8-12**	-0.06184	-0.03859	4.4E-08
5-6	0.01986	0.0355	4.5E-07	**6-14**	-0.0162	-0.0022	0.03161	**8-13**	-0.06929	-0.04796	3E-10
5-7	0.03405	0.04823	9E-11	**6-15**	0.0006	0.00912	0.06144	**8-14**	-0.02998	-0.01766	3.4E-07
5-8	0.03517	0.04934	4.9E-11	**6-16**	0.01053	0.01752	1.7E-07	**8-15**	-0.01227	-0.00717	4.4E-07
5-9	0.0356	0.04977	3.9E-11	**6-17**	0.01134	0.01832	6E-08	**8-16**	-0.00079	-0.00032	0.00028
5-10	-0.5795	-0.2887	2.2E-05	**6-18**	0.0117	0.01868	3.8E-08	**8-17**	0.000112	0.000383	0.00361
5-11	-0.1755	-0.0523	0.00381	**7-8**	0.00094	0.0013	1E-14	**8-18**	0.000473	0.000746	1E-09
5-12	-0.0214	0.00548	0.3254	**7-9**	0.00139	0.0017	2.2E-16	**9-10**	-0.62204	-0.33155	5.6E-06
5-13	-0.029	-0.0037	0.03456	**7-10**	-0.6205	-0.33	5.9E-06	**9-11**	-0.21783	-0.09534	0.00016
5-14	0.00919	0.02768	0.00151	**7-11**	-0.2163	-0.0938	0.00018	**9-12**	-0.06227	-0.03902	3.7E-08
5-15	0.02505	0.04002	1.1E-08	**7-12**	-0.0607	-0.0375	6.7E-08	**9-13**	-0.06972	-0.04839	2.6E-10
5-16	0.03462	0.0488	6.6E-11	**7-13**	-0.0682	-0.0468	4.6E-10	**9-14**	-0.03041	-0.01809	2.5E-07
5-17	0.03542	0.04959	4.3E-11	**7-14**	-0.0289	-0.0165	7.8E-07	**9-15**	-0.0127	-0.0076	2E-07
5-18	0.03578	0.04995	3.5E-11	**7-15**	-0.0112	-0.006	3.3E-06	**9-16**	-0.0012	-0.00076	6.2E-09
6-7	0.00997	0.01696	3.5E-07	**7-16**	0.00032	0.00081	0.00035	**9-17**	-2.81E-04	-8.12E-05	0.00363
6-8	0.01109	0.01807	8.2E-08	**7-17**	0.00121	0.00152	2.2E-16	**9-18**	8.05E-05	0.000283	0.00396
6-9	0.01152	0.0185	4.8E-08	**7-18**	0.00157	0.00188	2.2E-16	**10-11**	0.163926	0.476492	0.00138
6-10	-0.6071	-0.3165	9.1E-06	**8-9**	0.0003	0.00056	2.1E-06	**10-12**	0.280499	0.571798	2.9E-05
6-11	-0.2029	-0.0802	0.00049	**8-10**	-0.6216	-0.3311	5.7E-06	**10-13**	0.272157	0.563327	3.8E-05

Table 30: p-Values for End-to-End Delay (3 of 3)

Compare	Lower	Upper	P-value	Compare	Lower	Upper	P-value	Compare	Lower	Upper	p -Value
10-14	0.30719	0.59791	1.2E-05	**11-18**	0.09552	0.21802	0.00015	**13-18**	0.048572	0.069899	2.41E-10
10-15	0.32138	0.61191	7.8E-06	**12-13**	-0.0239	0.00711	0.369	**14-15**	0.007487	0.020716	0.00091
10-16	0.33057	0.62106	5.8E-06	**12-14**	0.01339	0.03941	0.0014	**14-16**	0.017105	0.029435	5.11E-07
10-17	0.33137	0.62186	5.7E-06	**12-15**	0.02863	0.05237	2.09E-06	**14-17**	0.017906	0.030231	2.82E-07
10-18	0.33173	0.62222	5.6E-06	**12-16**	0.03804	0.06129	5.39E-08	**14-18**	0.018268	0.030593	2.16E-07
11-12	0.04374	0.16814	0.00701	**12-17**	0.03884	0.06209	3.98E-08	**15-16**	0.006611	0.011726	1.19E-06
11-13	0.03548	0.15958	0.01213	**12-18**	0.0392	0.06245	3.47E-08	**15-17**	0.007415	0.012518	2.80E-07
11-14	0.07082	0.19385	0.001	**13-14**	0.02264	0.04697	1.68E-05	**15-18**	0.007777	0.01288	1.47E-07
11-15	0.08515	0.20773	0.00034	**13-15**	0.03798	0.05983	1.16E-08	**16-17**	0.000579	0.001017	4.33E-07
11-16	0.09436	0.21686	0.00017	**13-16**	0.04741	0.06874	3.71E-10	**16-18**	0.000941	0.00138	1.05E-10
11-17	0.09516	0.21765	0.00016	**13-17**	0.04821	0.06954	2.76E-10	**17-18**	0.000258	0.000467	2.96E-07

Table 31: p-Values for Packet Loss (1 of 3)

Compare	Lower	Upper	P-value	Compare	Lower	Upper	P-value	Compare	Lower	Upper	p -Value
1-2	2.126086	3.08108	1.44E-11	**2-6**	1.17396	1.71218	5.82E-12	**3-11**	-0.77569	-0.1738	0.01083
1-3	3.378065	4.294	2.20E-16	**2-7**	-2.6231	-1.7683	2.34E-11	**3-12**	0.429345	0.960001	5.14E-05
1-4	-0.314053	0.73368	0.5053	**2-8**	-0.197	0.58461	0.4101	**3-13**	-2.9059	-2.02374	1.06E-11
1-5	1.947148	2.93028	1.11E-10	**2-9**	1.07877	1.70953	6.53E-10	**3-14**	-0.72942	-0.17145	0.00914
1-6	3.5999	4.49342	2.20E-16	**2-10**	-1.7724	-0.8615	1.49E-05	**3-15**	0.645599	1.186256	5.04E-07
1-7	-0.144412	0.96021	0.2218	**2-11**	0.42649	1.08892	0.00032	**3-16**	-2.73118	-1.69108	1.73E-08
1-8	2.271833	3.32295	5.23E-12	**2-12**	1.62744	2.22681	2.53E-15	**3-17**	-0.77692	-0.0227	0.08175
1-9	3.523072	4.47239	2.20E-16	**2-13**	-1.6938	-0.7709	4.89E-05	**3-18**	0.45254	1.077509	0.00015
1-10	0.7130508	1.86015	0.00042	**2-14**	0.47035	1.09368	9.51E-05	**4-5**	1.822863	2.634943	9.71E-13
1-11	2.876474	3.8461	5.54E-15	**2-15**	1.84432	2.45243	2.20E-16	**4-6**	3.487907	4.185788	2.20E-16
1-12	4.065806	4.9956	2.20E-16	**2-16**	4.8081	5.78679	0.00383	**4-7**	-0.28121	0.677389	0.4924
1-13	0.7929238	1.9495	0.00021	**2-17**	4.8081	3.97546	0.00104	**4-8**	2.140055	3.035102	1.08E-13
1-14	2.913278	3.85792	5.40E-15	**2-18**	1.65586	2.33909	8.35E-14	**4-9**	3.403071	4.172772	2.20E-16
1-15	4.284374	5.21955	2.20E-16	**3-4**	-3.9897	-3.2627	2.20E-16	**4-10**	0.572663	1.580914	0.00075
1-16	0.9859428	2.26386	8.16E-05	**3-5**	-1.7097	-1.085	8.32E-10	**4-11**	2.753851	3.549102	2.20E-16
1-17	2.90326	3.96919	6.38E-15	**3-6**	-0.0189	0.44019	0.1305	**4-12**	3.948468	4.693319	2.20E-16
1-18	4.109415	5.0927	2.20E-16	**3-7**	-3.8332	-3.023	2.20E-16	**4-13**	0.65184	1.670965	0.00035
2-3	0.9442175	1.52068	2.11E-09	**3-8**	-1.4047	-0.6726	1.96E-05	**4-14**	2.793901	3.557673	2.20E-16
2-4	-2.78221	-2.0053	3.03E-14	**3-9**	-0.1215	0.44489	0.3436	**4-15**	4.166288	4.918008	2.20E-16
2-5	-0.50637	0.17663	0.4229	**3-10**	-2.9842	-2.1147	3.49E-12	**4-16**	0.836624	1.993552	0.00015

Table 32: p-Values for Packet Loss (2 of 3)

Compare	Lower	Upper	P-value	Compare	Lower	Upper	P-value	Compare	Lower	Upper	*p* -Value
4-17	2.770022	3.6828	2.20E-16	**6-12**	0.23986	0.72823	0.00164	**8-11**	0.163961	0.963835	0.02193
4-18	3.985112	4.79738	2.20E-16	**6-13**	-3.1048	-2.2461	1.37E-12	**8-12**	1.358404	2.108226	4.98E-10
5-6	1.312959	1.90293	6.61E-12	**6-14**	-0.9202	-0.402	8.42E-05	**8-13**	-1.93749	-0.91486	2.00E-05
5-7	-2.474072	-1.5876	4.09E-10	**6-15**	0.45564	0.95496	1.74E-05	**8-14**	0.203905	0.972512	0.01334
5-8	-0.049622	0.76697	0.1473	**6-16**	-2.932	-1.9115	2.30E-09	**8-15**	1.576249	2.33289	1.81E-11
5-9	1.22167	1.89637	2.01E-10	**6-17**	-0.9736	-0.2473	0.00725	**8-16**	-1.75247	-0.59251	0.00138
5-10	-1.622373	-0.6819	0.00015	**6-18**	0.25928	0.84952	0.00285	**8-17**	0.180464	1.097203	0.02333
5-11	0.5704949	1.27465	5.05E-05	**7-8**	1.90832	2.87067	2.14E-11	**8-18**	1.395276	2.212057	8.33E-10
5-12	1.769131	2.41485	3.29E-15	**7-9**	3.16569	4.01398	2.20E-16	**9-10**	-3.16357	-2.25869	5.28E-13
5-13	-1.543596	-0.5914	0.00045	**7-10**	0.34491	1.41249	0.00793	**9-11**	-0.96336	-0.30953	0.00191
5-14	0.6129776	1.28079	1.48E-05	**7-11**	2.51774	3.38904	1.54E-15	**9-12**	0.23811	0.827835	0.00375
5-15	1.986364	2.64013	2.20E-16	**7-12**	3.70978	4.53584	2.20E-16	**9-13**	-3.08502	-2.16801	1.70E-12
5-16	-1.363478	-0.2642	0.01672	**7-13**	0.42441	1.50222	0.00412	**9-14**	-0.91919	-0.30508	0.0015
5-17	0.5794017	1.41562	0.0002	**7-14**	2.55621	3.39919	9.45E-16	**9-15**	0.454916	1.053538	8.98E-05
5-18	1.800538	2.52415	3.22E-14	**7-15**	3.92797	4.76016	2.20E-16	**9-16**	-2.90755	-1.83811	3.89E-09
6-7	-4.03096	-3.2466	2.20E-16	**7-16**	0.6129	1.8211	0.00139	**9-17**	-9.59E-01	-1.64E-01	0.02185
6-8	-1.600877	-0.8977	4.82E-07	**7-17**	2.53897	3.51768	9.53E-15	**9-18**	0.26586	0.940788	0.00413
6-9	-0.312582	0.21473	0.7571	**7-18**	3.74982	4.6365	2.20E-16	**10-11**	1.611538	2.537838	1.39E-09
6-10	-3.18294	-2.3372	5.27E-13	**8-9**	0.81309	1.58759	3.57E-06	**10-12**	2.801985	3.686225	1.93E-15
6-11	-0.96816	-0.4026	0.00018	**8-10**	-2.0167	-1.0049	6.42E-06	**10-13**	-0.47614	0.645369	0.8017

Table 33: p-Values for Packet Loss (3 of 3)

Compare	Lower	Upper	P-value	Compare	Lower	Upper	P-value	Compare	Lower	Upper	*p* -Value
10-14	1.64903	2.54897	7.46E-10	**11-18**	0.88758	1.59196	2.13E-07	**13-18**	2.753669	3.706017	2.17E-15
10-15	3.020396	3.91032	2.20E-16	**12-13**	-3.6078	-2.7112	5.13E-15	**14-15**	1.070997	1.661725	1.75E-10
10-16	-0.285076	0.96167	0.3679	**12-14**	-1.436	-0.8543	1.52E-08	**14-16**	-2.29336	-1.22804	1.93E-06
10-17	1.635995	2.66325	3.74E-09	**12-15**	-0.0613	0.50386	1.96E-01	**14-17**	-0.34416	0.445408	8.31E-01
10-18	2.844119	3.7848	7.17E-16	**12-16**	-3.4319	-2.3797	2.21E-11	**14-18**	0.881435	1.549481	1.09E-07
11-12	0.8575377	1.4813	5.59E-08	**12-17**	-1.4802	-0.7088	1.86E-05	**15-16**	-3.65555	-2.59857	2.84E-12
11-13	-2.45915	-1.521	5.03E-09	**12-18**	-0.2526	0.39333	7.17E-01	**15-17**	-1.70472	-0.92676	7.80E-07
11-14	-0.299039	0.34766	0.9004	**13-14**	1.55832	2.47045	2.64E-09	**15-18**	-0.4779	0.176098	4.43E-01
11-15	1.074613	1.70673	8.58E-10	**13-15**	2.92962	3.83187	4.96E-16	**16-17**	1.224687	2.397961	3.88E-06
11-16	-2.280073	-1.1927	3.05E-06	**13-16**	-0.374	0.8814	5.02E-01	**16-18**	2.426427	3.525887	1.12E-11
11-17	-0.335036	0.48491	0.7608	**13-17**	1.54605	2.58397	1.28E-08	**17-18**	0.746636	1.583031	2.07E-05

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14. ABSTRACT

Mobile ad hoc networks (MANETs) have grown in popularity over the past decade and are increasingly considered for time-sensitive multimedia applications. The impact of various routing protocols on voice traffic using different IEEE 802.11 extensions has been investigated via analytical models, simulations and experimental test beds. Many studies determined that optimized link state routing (OLSR) is a suitable routing protocol to support voice over internet protocol (VoIP) conversations. This research expands upon this understanding by determining the point at which voice traffic is no longer feasible in an ad hoc environment and determines which audio codec is best suited for MANETS.

The MANET simulation environment is established using OPNET. Varying combinations of workloads are submitted to the MANET to capture voice performance within a stressed environment. Performance metrics are compared against established benchmarks to determine if thresholds for unacceptable voice quality are exceeded.

Performance analysis reveals that VoIP communication using G.711 is not sustainable at walking (1.5 m/s) or jogging (2.5 m/s) speeds when three simultaneous streams are used. Also, G.729a is determined to be the best suited codec for MANETs since it significantly outperforms the other codecs in terms of packet loss and end-to-end delay.

15. SUBJECT TERMS

Mobile Ad Hoc Network, Voice Communication, Wireless Networks, Optimized Link State Routing, OPNET Simulation

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